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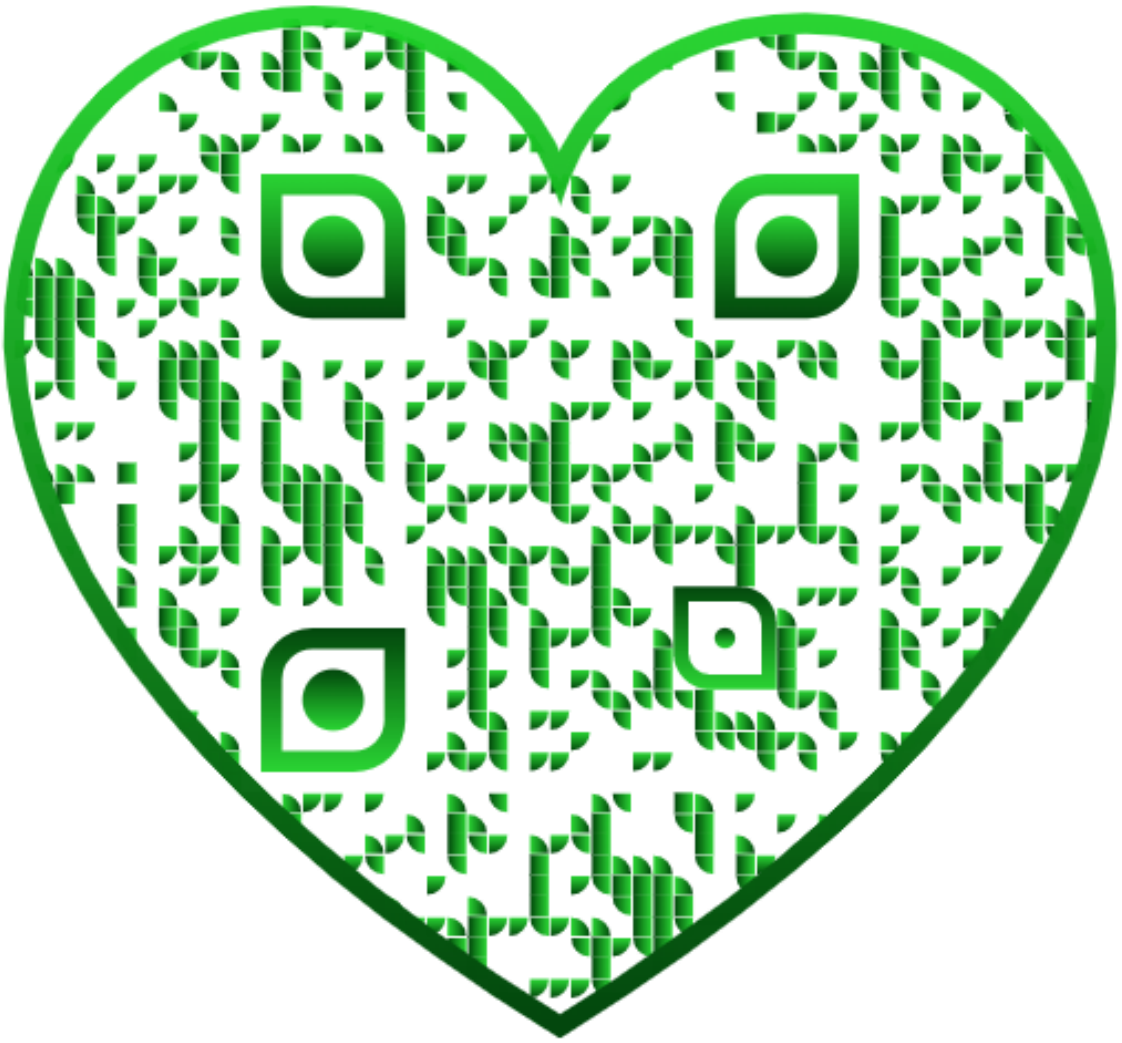

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AWARD WINNING

The greatest technological leaps forward are the ones we later come to take for granted. The Internet. Mobile telephony. Mains electricity. And, if you're a musician, the MIDI protocol, which celebrates its 40th birthday this year.

Nowadays, we barely give a second thought to the idea that a keyboard or fader surface might simply be a dumb controller, instructing something else as to how to produce a sound. After all, conceptually, it's not so very different from using a QWERTY keyboard to address a computer. But back in 1983, this itself was a radical concept in a world where most people still used typewriters. Even more outlandish was the notion that different instruments from different manufacturers might acknowledge a common control protocol.

The fact that MIDI achieved widespread adoption in the first place was a minor miracle. The fact that, four decades on, we still use it all the time is a testament to the foresight of its designers. Few of us still plug in 5-pin DIN cables or hook up hardware sequencers, but it's still fundamental to the operation of our studios and to our creative practices, to the point where it's hard to even imagine an alternative reality where MIDI didn't catch on. The magazine you're reading probably wouldn't exist if it wasn't for MIDI and the innovations it has enabled.

Even now, inventors and software developers are finding new ways to exploit the expressive potential of MIDI, and MIDI 2.0 is poised to unleash a new world of possibility. So when we were approached by the MIDI Association to ask if we'd like to be official media partners for the MIDI Innovation Awards, we jumped at the chance.

Now in their third year, the Awards aim to celebrate and reward creative and original applications of MIDI from hardware designers, software programmers and artists. The Awards attract many entrants from all over the world, who stand to gain not only invaluable publicity but also tangible help for their enterprises. Winners in each of five main categories are chosen by a stellar jury from a shortlist compiled through public voting — and that's where you come in!

The voting process is designed to be easy, fun and informative. Head to the *Sound On Sound* website or www.midi.org/innovation-awards to learn about the entries, watch them in action, and cast your votes. Last year's finalists ranged from face-powered synthesizers and knitted keyboards to Bluetooth sensors, iOS synths and chord-bending plug-ins. What new applications for this venerable but still revolutionary technology will this year's entrants find? I, for one, can't wait to find out.

Happy birthday, MIDI!

Sam Inglis
Editor In Chief

"The magazine you're reading probably wouldn't exist if it wasn't for MIDI and the innovations it has enabled."

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Sony MDR-MV1

Open-backed Headphones

Sony's first open-backed headphones offer an intriguing alternative to the established classics in the field.

SAM INGLIS

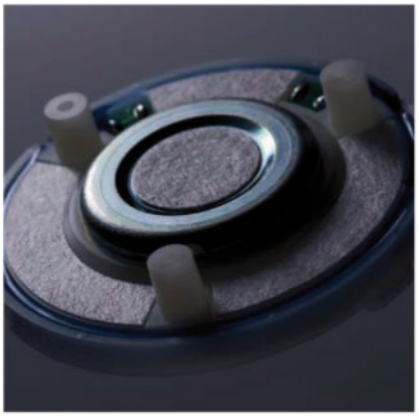
Although their products sometimes fly below the radar in the UK, Sony Professional have an enviable track record when it comes to making headphones. My own MDR-7509s were the first good-quality pair of cans I ever bought, nearly 25 years ago, and they're still going strong today. However, whereas some manufacturers have a sonic signature that is common to all or most of their models, Sony designs tend to do their own thing. The two closed-back models in the current catalogue are the MDR-7506 and the MDR-7510, and to my ears, they sound quite different from one another. As far as I know, moreover, Sony have never before made a pair of open-backed headphones — so what should we expect from the new MDR-MV1?

The Master Plan

Sony's promotional material claims an enormous frequency range, but as usual, no tolerances are given, so this information is of limited value. More intriguing are the suggested applications. Sony say of the MDR-MV1 that: "Their open-back design enables accurate reproduction of a wide sound field, making them a suitable option for mixing and mastering immersive spatial sound." Given that Sony's 360 Reality is embroiled in a hard-fought struggle for market supremacy against Dolby Atmos, it makes sense that they would want the tools for creating it to be as widely available as possible. One would imagine that any given pair of phones would be equally good or bad for both formats, of course, but the idea of headphones optimised for immersive mixing is an interesting one.

The marketing blurb also places great emphasis on the MDR-MV1s' light weight and comfort. At just 223 grams (about 8 ounces) they're certainly in the featherweight category; in fact, I don't think I can ever recall testing a pair of headphones that sat so lightly on the head. Comfort is more subjective, but they ticked all the right boxes for me, thanks to the soft removable earpads and generously padded headband. As on many modern headphones, the earcups are quite elongated, and should fully surround all but the largest ears. Hinged joints allow smooth and free rotation both fore and aft and up and down. The removable cable attaches to the left earcup using a TRRS mini-jack with a screw-on barrel. An adaptor is included to convert the quarter-inch jack at the other end to a mini-jack.

The flip side of the slimming diet is that nearly all the component parts of the MDR-MV1s are made out of hard »



The MDR-MV1s use a new, specially developed moving-coil dynamic driver that is said to give them a frequency response of 5Hz to 80kHz.

m908. Atmos® Music, Simplified

The m908 Monitor Controller gets you working in any format quickly and easily, expertly managing all speaker systems from stereo to Dolby™ Atmos 9.1.4. New firmware 2.0 includes a web-based control platform which lets you operate and configure the entire system from any web browser (desktop or mobile). Additionally, the m908's room correction EQ capability has been tweaked to provide 12 bands on all 24 channels at all sample rates. These updates, combined with its unrivaled audio performance and mechanical elegance, cement the m908 as the finest, all-in-one monitoring tool for modern music production.

"I find a monitor controller to be an indispensable part of studio life these days, particularly in a console-less room without a master section for speaker/input switching, and now with multi-channel mixing like Atmos becoming more prevalent, a properly built monitor controller for that scenario is vital. The Grace Design m908 absolutely fits that description; incredible sound quality, options for almost any input and output need, the flexibility to program multiple speaker layouts and have them available at the press of a button are just a few of the top features in a truly full-featured unit."

Don Gunn, TapeOp Magazine

GRACE DESIGN

www.gracedesign.com

The removable TRRS cable locks into the base of the left earcup using a threaded barrel.

plastic rather than metal. There's no reason to think this should make them less durable, and Sony make a point of emphasising their "long-lasting wear and reliability", but in terms of look and feel, there's little to separate them from headphones costing a third of the money. That perception is reinforced by the fact that Sony don't include any sort of a case, or even a soft bag, in their commendably sustainable paper-based packaging.

Like most of today's headphones, the MDR-MV1s are a low-impedance design (nominally 24Ω, according to the specs), and they offer a generous 100dB/mW sensitivity. I had no trouble driving them loud from low-power sources such as laptop headphone sockets.

Surrounded

Returning to Sony's marketing literature once again, it's notable that the emphasis is on accurate spatial presentation within a stereo and surround soundfield, rather than on presenting the most neutral frequency response. When mixing or consuming immersive audio on headphones, what we're hearing is a downmix of the spatial mix that's been processed through a binaural encoder. One complaint I've heard about this process is that the encoded mix can feel a bit hollowed-out in the low midrange.

I'm not sure whether Sony's engineers share that perception and have tried to correct for it, but the MDR-MV1s sound as though that could be the case. With a definite lift somewhere around the 100Hz region and a prominent treble boost above 5kHz or so, their sound is perceptibly 'scooped'. In fact, auditioned with stereo material, they were both brighter and bassier than any of the other open-backed phones I had on hand for comparison. But perhaps that's exactly what is needed for immersive mixing?

To find out, I auditioned Sony's own 360 Reality demo on multiple headphones. The binaural encoding sounded pretty ropey on all of them, with obvious phasey artefacts, but I did feel that the MDR-MV1s' 'smile curve' tonality improved matters somewhat. The extra high-end detail made it easier to localise objects within the 360-degree panorama, while the low-mid boost helped to restore some weight in an otherwise thin sound. Personally, I don't think that it's desirable to mix immersive audio wholly or mainly on headphones, but if you have to do it, I can see why Sony feel the MDR-MV1s are suited to the job.

For stereo work, I think the treble boost is something the ear will adapt to fairly easily, and many engineers value the detail that such an enhancement brings. I found it a little harder to get a handle on the low end, not so much because of its prominence as because it sometimes seemed rather woolly. One track I quite often use for auditioning headphones is 'Wire' from Jon Hopkins' album *Insides*. Some cans can't give the descending bass line the weight and power it needs, while others don't resolve the musical information clearly. The Sony cans passed the first test with aplomb, but I was less convinced by their ability to discriminate pitch, and there was something of a 'one note' quality to the results.

All in all, the MDR-MV1s present an interesting contrast to other open-backed headphones. For a similar outlay, models like the Neumann NDH-30 and Shure SRH1840 have a frequency balance that, to my ears, is about as subjectively flat as you'll get from any monitoring system. Sony's designers seem to have pursued a slightly different goal here. To turn the MDR-MV1s into a completely neutral monitoring tool for analytical listening, I think you'd need to pair them with custom equalisation from a product like Sonarworks or RealPhones — but if you are one of the many engineers who likes a richer low end than most headphones deliver, these could be right up your street. And if you're looking to get into immersive mixing and can't justify a five-figure speaker setup, they might just get you a touch closer than more conventional headphones.

summary

With the MDR-MV1s, Sony seem less focused on tonal neutrality, and more on other qualities that will help engineers mixing immersive audio on headphones.

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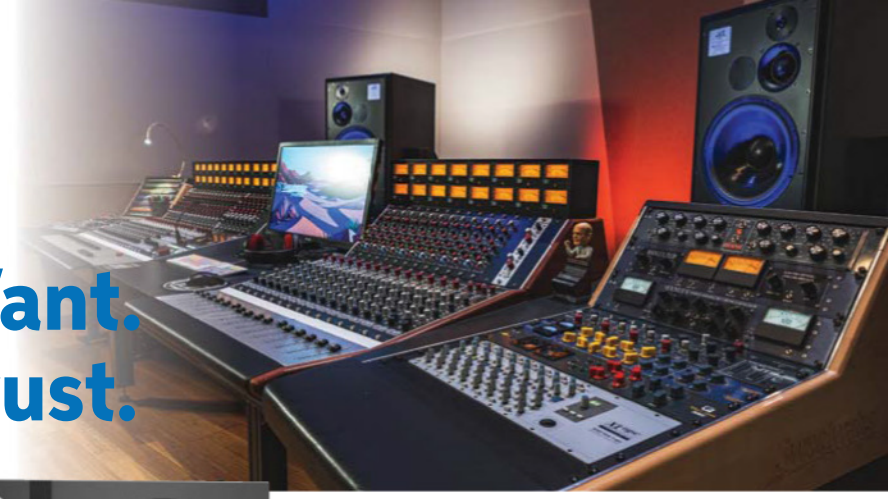

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Black Lion Audio Auteur DT

Microphone Preamplifier & DI

There's more to this desktop preamp than turning up your mic!

BOB THOMAS

Chicago-based Black Lion Audio forged their reputation with a range of performance upgrades to other manufacturers' audio interfaces. Work in this area continues, but over the years BLA have introduced their own original products, including the Auteur range of transformer-balanced mic preamps. The first Auteur, a dual-channel preamp, was released in 2009, and MkII and MkIII versions now sit alongside the four-channel Auteur Quad and single-channel Auteur MkII 500 variants, the latter being a 500-series module. The latest addition to the family is the Auteur DT, a single-channel microphone preamplifier/DI that follows its predecessors' transformer-balanced paradigm, but which has been specifically designed for use on a desktop.

Catching The Eye

The DT looks attractive, and is compact and surprisingly heavy, with its black-painted, brushed-aluminium chassis (which features "vintage style, translucent maple side panels") and a hefty analogue 24V AC external power supply contributing to a total overall weight of 1.34kg. The channel controls are the same as on all Auteur models: a rotary 0-60 dB gain control, and buttons (joined by status LEDs) for 48V phantom power, polarity inversion, a -10dB pad, and to engage the Hi-Z input. That last button toggles the input source between the rear-panel balanced XLR mic input and the front-panel high-impedance quarter-inch TS jack DI input. A backlit, retro-styled VU meter displays the post-gain, post-pad signal level, and a power on/off switch completes the front panel. On the rear, you'll find the XLR mic input, a quarter-inch TRS jack for the transformer-balanced output, and the connector for the power supply.

Internal components, which include Vishay and Nishicon capacitors and an Edcor PC 600/600 steel-cored output transformer, are of high quality, as are the PCB layouts and internal construction. Like the rackmount Auteur MkIII, the DT features what Black Lion Audio describe as a "modern IC-based front end for fast and transparent sound" and a "vintage-inspired, transformer-coupled output stage for massive, colourful sound". Other than quoting an audio bandwidth of 10Hz to 30kHz, Black Lion Audio are a little coy about the technical specifications, though those for the related Auteur MkII 500 suggest that the specs should be good. The unit has a 600Ω output impedance.

On Test

Sonically, the Auteur DT delivers a clear, precisely delineated (but definitely not clinical) response that exhibits the low-end warmth, smooth midrange and detailed high end that characterise a high-quality microphone preamplifier with a transformer-balanced output. Provided that you avoid overloading the inputs of your audio interface or mixing console, you'll find it very difficult to extract a poor performance from this preamp. Vocals, acoustic instruments and percussion all benefit from the additional low-end weight and warmth imparted by the DT's Edcor output transformer.

The DI input is capable not only of adding richness and depth to electric guitars and basses, but also of enhancing the performance of hardware-based amp modellers and loudspeaker IR loaders. For example, the Auteur DT brought a distinct improvement to the overall sound of my Amplifire and Strymon Iridium amp and cab simulators, especially when I pushed the levels a bit, and I'm now extremely tempted to keep those combinations available on a permanent basis.

The only operational drawback that I could find in the Auteur DT is the intrusion of self-noise at very high gain control settings, which could become an issue when using low-output passive microphones.

Overall, then, the Auteur DT is an impressive, versatile and compact microphone preamplifier and DI. Sonically, it punches well above its relatively modest price, and its size, appearance and portability just add to its attractiveness. I can see this preamp appealing especially to musicians, podcasters and voiceover artists looking for a high-performance single-channel microphone preamplifier with real character that won't break the bank — and will look great on your desk! If you're in that group, I recommend giving the Auteur DT a very close look and listen.

ALTERNATIVES

If you've got a bit more, or a bit less, to spend, you'll find the more utilitarian-looking **Golden Age Premier Pre-73 Jr MkII** and **Fredenstein VAS** single channel, transformer-equipped microphone preamplifiers sitting, respectively, at price points above and below that of the **Auteur DT**. Other than **Black Lion Audio**'s own closely priced, half-rack **B12A MkIII**, you'll need to spend a good bit more to find other options offering similar specifications.

summary

An impressive, versatile and compact single-channel, transformer-equipped microphone preamplifier and DI that, in terms of sonics, manages to punch well above its relatively modest price.

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Orchestral Tools Salu

Sample Library

Orchestral Tools get up close and personal with some intimate soundscapes.

MARK NOWAKOWSKI

Orchestral Tools have recently been stretching out into new sonic territories, with their offering Salu, subtitled Close-up Soundscapes, being the more adventurous companion to last year's luminous Tallinn library. While Tallinn brought us the evocative and ethereal sounds of the world-class Estonian Philharmonic Chamber Choir, Salu takes us to the Arvo Pärt Centre recording studio to explore more intimate and experimental musical textures. The library features solo strings, a string quartet, a folder of interesting and sometimes gorgeous voice instruments, keyboards, plucked instruments, percussion, and a final folder of processed interpretations. The recordings are dry and close, allowing for great customisation in the production process.

Collected Sounds

Beginning in the Strings folder, we encounter a String Quartet instrument capable of playing lush legato passages, stacked sustains, doubling at octaves, and a host of techniques from portatos to swells and even maj/min arpeggios. The ensemble unison sometimes limits the range, but the instrument is agile and plays beautifully. While a solo violin is not included, an effective solo viola and solo cello patches round out the capabilities of this folder. There is an unevenness of dynamics at times — such as the top dynamic layer of the sustain patches being outgunned by the volume of the arpeggio patches — but these are small concerns next to the deeply personal sound of these patches. Close and intimate is the key here, and the dynamics of the performances are tastefully restrained. A particular favourite from the solo instruments were the circular bowing patches.

The Voices folder contains male and female ensembles, a folder of extended techniques, along with solo alto and basso voices. Beginning in the Female Ensemble, each instrument is organised into unison legatos, sustain patches, different syllables, crescendo lengths, hums and swells. The solo voices sound great as well, although some odd chorusing and smear was encountered in the legato patch. Doubtless this will be addressed in a future update. As to the special articulations folders, they contained a number of techniques to supplement a standard choral instrument, with the irregular repetitions and cascading vowels a particular highlight of the folder.

A real highlight of the collection is the Plucked & Keys folder. The unison kannel/harp/piano patch has a breathtaking and shimmering immediacy; from the irregular repetitions to the minor-key arpeggios, there is much to like here. The three instruments are also sampled individually: the solo kannel open arpeggios patch was full of soundtrack moodiness, while the irregular harp repetitions and octave drops sound great layered. The excellent solo piano is dark and moody, with rich and gorgeous irregular repetition and arpeggio patches.

Perhaps the most surprising aspect of the collection is the Percussion folder, which contains multiple techniques sampled across 11 instruments. Highlights include the sonorous soft-mallet tubular bells, a velvety vibrato induced by dipping the bells into water, glock and crotale bends, bass drum played with a rubber ball, a vibraphone with random vibrato for each note, and a folder full of amazing thundersheet sounds: I predict the sound of bowed thundersheet will be popping up in soundtracks all over the place. It's an exceptional collection which can deeply supplement any standard percussion library.

Finally in terms of sound design, the various synthesized textures emerging from this library provide some sound-design glue to go between the natural recordings. Just go into the Evolving folder, for instance, and pick Upright Dreaming for instant sparkling piano accompaniments, or Tranquil Tides for an immediate evolving plucked pad for your next ethereal track.

Conclusion

In the final analysis, Salu is a feature-rich, varied and frequently beautiful library capable of richly supplementing existing standard instrumental libraries. If Orchestral Tools have made two mistakes, it's in underselling the depth of their latest collection, and also not providing a convolution reverb to allow greater sonic matching with the Tallinn library. While the $399 price tag is appropriate for a library of this magnitude, the option to purchase individual instruments for as little as $22 is a more affordable way to sample these highly satisfying and musical virtual instruments.

summary

Orchestral Tools Salu provides an intimately detailed set of musical tools to deeply supplement more standard instrumental libraries. The instruments are musical and highly playable, containing many surprising sounds. A few minor hiccups in execution will hopefully be ironed out in updates.

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Malcolm Toft Punishr

500-series Distortion & EQ

Controllable carnage is the name of the game, but the Punishr can do warmth and subtlety too!

NEIL ROGERS

Due in no small part to his having developed the sought-after Trident consoles in the '60s and early '70s, Malcolm Toft is rightly seen as a legendary figure in the pro audio industry, but he's also been the man behind various other console and outboard brands over the years. Most recently he's begun developing designs under his own name.

During the Covid lockdowns, Malcolm was approached by London-based Engineer Nick Mitchell about an idea for an 'analogue distortion box', conceived as a high-quality alternative to the many saturation and distortion plug-ins out there. Enthused by the idea, Malcolm worked up the design and, after a period of experimentation and beta testing, the fruits of their collaboration were released in the form of the Punishr Analogue Harmonic Distortion Module. It's an all-analogue affair that manages to pack not only three different types of distortion but also a fully fledged EQ section into the confines of a single-width 500-series module. I was sent a pair of these modules for review, to enable me to use them on busses and other stereo material, and was very keen to discover what analogue punishment they could dish out in my studio!

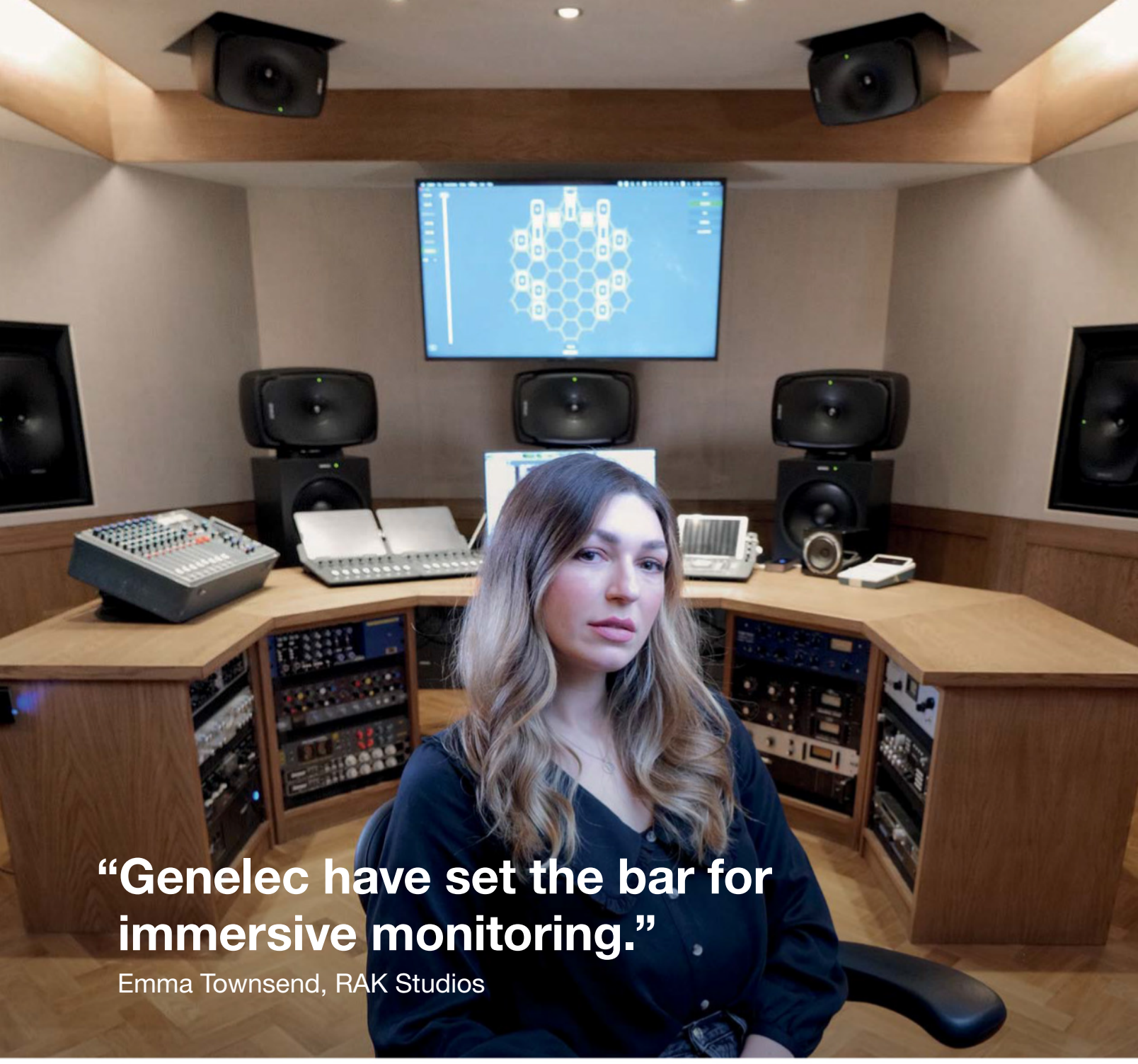
Dishing The Dirt

The Punishr's distortion section offers three different stages for 'attacking' your audio, and each has both a bypass button and a knob to drive that stage harder. The first is described as Symmetrical distortion and introduces clipping to both sides of the waveform. This results in a fairly conventional (but not unappealing) sort of distortion effect, and I generally found this to be a great place to start experimenting with most material if I didn't already have specific plans for it. Things get a bit more wild with the Asymmetrical stage, which introduces clipping to only one side of the waveform — this is definitely the unruly sibling of the trio! Lastly, we have Iron, which as the name suggests employs a small custom transformer. It was chosen for no operational reasons other than how it sounds when it distorts, and how nicely its effect combines with the other two distortion types.

By default, in what will be the normal mode of operation for this section, the audio signal passes through each of these three distortion sections in series, but in the EQ section below (of which more later) there's also a Sum button. Press this and the input signal is fed in parallel to all three distortion sections, and the output is then summed before going on to feed the next stage in the module. It's also worth mentioning that each distortion section has an automatic gain compensation facility to ensure the signal doesn't increase in volume significantly as more character is added. The idea, of course, it to make it easier to make judgements about the tonality, without level changes skewing your perception. It's a hugely useful feature and while we may be used to seeing it in plug-ins, it isn't commonly found in hardware.

Beneath the distortion section is the EQ. Malcolm Toft obviously knows a thing or two about designing EQ circuits, and while this is a compact EQ it's been well thought out and adds a huge dollop of flexibility and value. First, there's a fixed-frequency shelving EQ (±15dB of gain at either 100Hz or 8kHz) that allows you to make broad changes to the tonal balance. Below this a 12dB/oct variable high-pass filter that can be set from 35Hz to 350Hz, and a low-cut filter that can be swept from 1.5kHz up to 20kHz. The filter and shelving stages can be bypassed separately.

With all the distortion stages bypassed you could, if you wished, use this simply as an EQ, which is handy. But the real reason it's here is to shape your distortion. If you're a regular user of saturation/distortion style effects, you'll know just how much difference it makes if you place an EQ before or after the distortion stage, and Punishr gives you the ability to toggle the EQ section pre and post distortion. »



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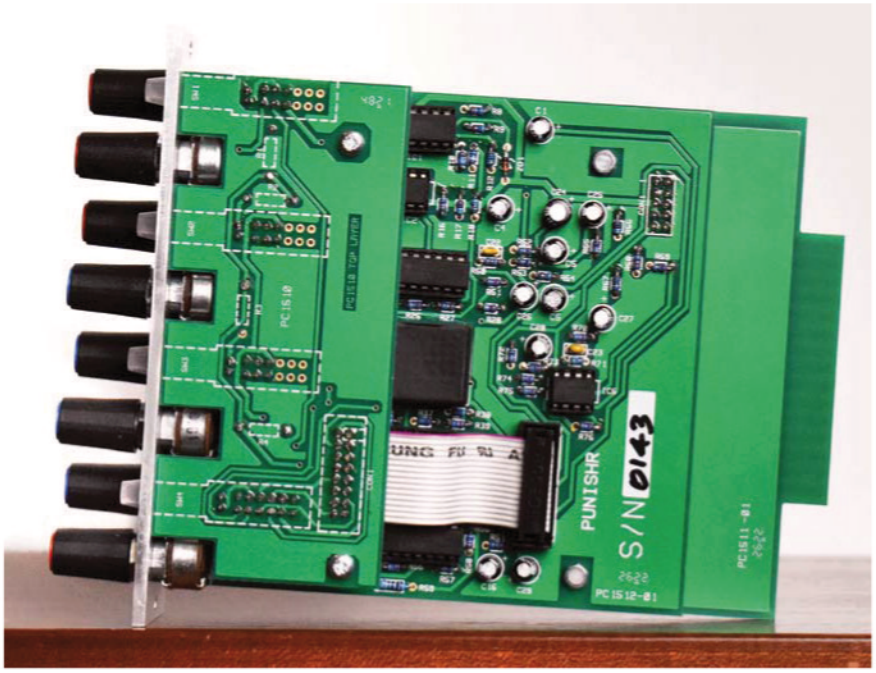
» The last feature of note, which comes at the end of the signal path, is a dry/wet blend control — it's a welcome addition to any analogue processing device, but hugely significant here, as it makes it much easier to coax usable results from the more full-on options available in this device.

In Use

There are many positives to the 500-series format but one of the downsides is that the small front panels mean controls can often feel very crowded. I talked to Nick Mitchell to check a few facts as I was finalising this review, and he explained that a lot of thought had gone into the layout and the choice of pots used, the main idea being to ensure that the Punishr felt like using an intuitive console-style channel strip. This care seems to have paid off, since while the Punishr does pack a lot of controls into the available panel space, I was pleasantly surprised by just how quickly I was able to get my bearings — it felt like a matter of minutes before I was playing with sounds and not even thinking about what control was where.

The bread and butter of my recording and mixing work involves 'real' instruments played live, which usually demand more subtle saturation treatments. The Punishr can certainly deliver the sort of warmth or edge required for that but, to cut a long story short, I found it to be most effective when used with loops and more programmed material like electronic drums, synths and Moog bass lines. Putting a drum loop through this thing is straight-up fun, and from crunchy distortion through to full-on glitchy annihilation it felt both inspiring and refreshing to be indulging in this kind of tomfoolery outside of the box. Fuzzed-out bass sounds were another favourite subject for abuse, as was anything that suited a dose of fizzing, upper-midrange excitement.

For more conventional mixing duties the Punishr also has lots to offer, and one effective use for distortion in a mix is to inject a sense of 'definition' or 'forwardness' into vocals or certain other parts. The EQ section really comes into its own here, and is the key to your ability to go beyond 'creating cool sounds' and on to actually getting things to work in a mix. The combination of the two filters and the high and low shelving EQs was perfect in this regard, and I loved the hands-on immediacy of being able to tame a more 'out there' distortion effect with a nice analogue EQ. The regular EQ can mellow



Traditional through-hole components are used throughout this all-analogue design.

things out if needed, then, but the Sum option also tends to have the effect of calming things down and creating more mix-friendly effects — I suspect this is due in large part to how the transformer reacts to the two waveform clipping options. In practice, I tended to find that I engaged the module on a source and had fun taking things a bit too far, before using a combination of the wet/dry control and EQ to shape it to suit the mix.

Summing Up

The first thing I had to learn about the Punishr was what it *wasn't* trying to be. I came into this review off the back of assessing some very high-end valve-based outboard gear that imparted a richly colourful saturation to any audio passing through it. This isn't at all what the Punishr is about! While you can coax subtle, vintage-style effects from it using a combination of the Symmetrical and Iron controls in parallel, this just isn't the unit's reason for being. Rather, it's an analogue tool for creating bitey, nasty, fizzy distortion that transforms a sound, and then offers you the tools to caress and cajole into working in a mix.

When I first read the Punishr marketing material, my initial thought was that it was explaining the functionality you'd expect from a well-designed plug-in — Malcolm Toft really has done a superb job of executing all of this in the analogue domain. The ability to bypass each section and the gain levelling circuitry make it so easy and enjoyable to audition different sounds, and for many DAW-based mixers trying this out, it's going to feel like a novelty getting your hands dirty when playing with this style of hardware.

It's not just about the functionality, however. To my ears, the Punishr sounds different in style and flavour from most of the well-known distortion plug-ins, which so many of us now tend to use. The quality of the EQ section is a bonus too — while it's a crucial tool for shaping the distortion effects, it can hold its head high as a standalone EQ.

I had a lot of fun punishing my audio with this well thought out 500-series module. It can be used for anything from subtle to crazy effects — it's an excellent tool for shaping and creating sounds at the writing stage, or for adding personality when mixing — but it should appeal particularly to those working with synths, drum loops, and programmed material in general.

summary

The Punishr offers a creative and creditable hardware option for distortion effects, whether you're writing, producing or mixing music. With three distortion flavours and a great-sounding EQ section, the Punishr is both fun and good value.

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Extinct Audio BM9A

Active Ribbon Microphone

Extinct Audio bring some fresh activity to their popular ribbon design!

SAM INGLIS

If you want to know how good a piece of gear is, there are lots of questions you can ask. How does it perform on the test bench? What records was it used on? How does it sound in shootouts? Did that bald guy from *SOS* like it?

These are all valid criteria, especially the last one. But I'd like to propose another, under-used metric. How often to people ask to borrow it? My mics enjoy plenty of action with friends and colleagues, and it's no surprise that lots of people like to have a U87 on hand for a vocal session, for example. But the Neumann classic still comes a distant second in my league table of most-borrowed mics.

Nine Lives

In *SOS* April 2018, I reviewed the Extinct Audio BM9 Viking, a passive ribbon microphone inspired by an old Bang & Olufsen design. I bought the review pair, and they've rarely been back in their boxes since. The BM9s get borrowed at least twice as often as any other mic, because they sound great on almost everything: drum overheads, guitar amps, brass, room miking, even lead vocals on occasion.

In fact, Extinct got so much right with the BM9 that all their subsequent products have essentially been variations on it. There's the ruggedised Black Ops, the stereo Valkyr, and now the BM9A, a phantom-powered version with a built-in active buffer/gain stage. It's outwardly identical to the passive version apart from the '48V' on the name badge (owners of both models might want to do something more obvious to differentiate them!). And, according to Extinct founder Stewart Tavener, it should sound exactly the same, except that it's 22dB more sensitive.

Out & About

In a passive ribbon microphone, the transformer acts as an impedance converter, provides 'free' gain to boost the minuscule signal level coming from the ribbon element, and also balances the output. In an active mic, the output is electronically balanced instead, and this leaves more choices open relating to the transformer. Assuming part of the reason for making the mic active is to raise sensitivity, you can retain the same transformer and include a gain stage in the active electronics; or you can use a transformer with a higher turns ratio, with the active electronics being simply a buffer that stabilises the output and allows it to drive long cable runs. Or you could combine the two approaches.

Getting lots more gain from the transformer alone, though, requires a complete redesign using many more coils of thinner wire, increasing its capacitance and potentially changing the sound of the mic. Consequently, the active BM9 uses exactly the same custom transformer as the passive version, designed and built in-house, but married to an active output stage that provides 22dB gain. It's like having a passive BM9 with a Cloudlifter built in, and it does sound identical to the passive BM9 — with one caveat.

Transformer-balanced mics can have a slightly symbiotic relationship with the preamp they're feeding. A passive BM9 through a Neve 1073 might not always sound the same as a passive BM9 through an API 512, and so on. By contrast, the electronically balanced output stage in the BM9A should be relatively indifferent to preamp and cable loading. It's also, of course, 22dB hotter, which can make a difference if you like to drive preamps hard and get a little saturation into the sound.

So, when I lent the active BM9s to a friend who often borrows the passive ones, he reported that they didn't quite sound the same through his UA 610s, and I too managed to make them sound different by deliberately overdriving a vintage mic preamp. In the normal run of affairs, though, what you're getting is exactly what it says on the tin: a BM9, just louder. It wasn't broke, they haven't fixed it, and it's still a great mic!

summary

Unsurprisingly, the active version of Extinct Audio's excellent ribbon mic sounds like the passive version — but it's more sensitive.

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IK Multimedia Tonex Pedal

Amp Profiling Pedal

The Tonex Pedal makes IK's powerful amp-capture technology available in a standalone hardware unit.

PAUL WHITE

When I reviewed the IK Multimedia Tonex software (*SOS* January 2023: www.soundonsound.com/reviews/ik-multimedia-amplitube-tonex), I was impressed by its performance in the studio, but I also commented that a pedal version that allowed the amp models to be used live would be more than welcome. Well, it seems I wasn't the only one to have that idea: IK Multimedia must have already been working on it because in March they announced the Tonex Pedal!

Overview

The Tonex software, which is included in the price when you purchase the pedal, uses machine learning to capture the performance of a miked guitar amplifier, pedal or amplifier/pedal combination at specific drive and tone settings, in a similar way to other amp-profiling devices. IK call the captured amp a Tone Model. The types of pedals that can be modelled include drives and EQs, though it's worth noting that modulation and delay effects can't be captured. The process of capturing an amplifier can take up to half an hour or so because there's a lot of number-crunching going on here, but to my ears, as long as you're careful to set up your levels correctly, the results are virtually indistinguishable from the real thing. The software helps to make this easy, by offering prompts, metering and instructions throughout the procedure.

For those unfamiliar with Tonex or, indeed, any other amp-capture technology, what you capture is the amp's sound at your current settings. It can't capture the whole range of sounds of which the amp is capable, or at least not in a single Tone Model, but there is nonetheless some leeway to adjust the drive and EQ on the pedal or using the Tonex software. If you need both clean and driven sounds from »

IK Multimedia
Tonex Pedal
$399

PROS
- Tone Models can be taken on the road with no computer!
- Intuitive controls.
- Holds 150 presets.
- Tone Models have a very authentic sound and playing feel.

CONS
- No amp tremolo.

SUMMARY
IK's Tonex software was already a powerful amp-capture system, and now with this pedal, you can use it without a computer.

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» the same amp though, for the best results you'd make a separate Tone Model capture for each.

The advantage of the new pedal is that your choice of Tone Models can be loaded into the device as presets, which are then ready to use on stage or in the studio, without having to hook your guitar up to a computer. The pedal hosts both amp and speaker cab Tone Models, and that means you can plug the pedal directly into a PA or full-range amp. You can, of course, bypass the cab modelling if you prefer to plug it into your guitar amp — if your amp has an effects loop return, plugging the pedal into that rather than the regular guitar input will get you the most accurate sound, as the Tonex pedal can then be fed directly to the power amp and speaker, bypassing the preamp and tone stacks.

While the Tonex software gives you the opportunity to capture the character of real amps, whether your own or those you hire or borrow for the purpose, you don't have to rely on self-captured models — you also get access to a large library of already captured amp and pedal Tone Models that can be transferred to your pedal. Purchasers can choose from 1000 premium Tone Models via the Tonex software's Library section, plus an unlimited number of User Tone Models on ToneNet. Up to 150 Tone Model presets, arranged as 50 banks of three presets, may be loaded into the pedal. As supplied, these are already populated with a variety of amplifier types, from super clean to extra driven, but any of these can be overwritten with an alternative preset. Tone Models may also be accessed from the free Tonex iOS app and the free version of Tonex.

IK's Tonex Max and AmpliTube 5 software can be download once the pedal is registered, and in addition to using Tonex to load your pedal, you can use it either as a plug-in compatible with all the common formats for Mac and Windows DAWs, or hosted within AmpliTube 5. Used within AmpliTube 5, Tonex significantly expands the range of amp types that are available to use in your signal chain.

Now, you might be thinking at this point that if you've already bought the Tonex software (or AmpliTube 5, come to that), paying the full price for the pedal plus software bundle seems a tad unfair. Well, fear not: registered owners of Tonex Max software can get $100 off the Pedal via the IK Multimedia online store by using the auto-coupon that appears when you go to checkout. Users of other IK products, such as AmpliTube, will have received IK JamPoints from their purchases, and these too may be used to get a discount off the full Tonex Pedal if bought through the IK website.

As with many digital pedals these days, the encoders and footswitches can perform multiple functions.

Layout & Connectivity

The Tonex pedal uses the same aluminium housing and control setup as the company's X-Series effects, with eight knobs and three footswitches plus a large display area showing preset name and bank number or parameter name and value. The A-D/D-A converters are 24-bit/192kHz devices and allow the pedal to boast a huge 123dB dynamic range, and a frequency response of 5Hz to 24kHz. As with the existing X-Series pedals, the Tonex pedal has a USB port that allows the pedal to be used as an audio interface (at a fixed sample rate of 44.1kHz), and which also allows it to communicate with a device running the Tonex software.

On the front panel are turn-and-push encoder controls to select the model, preset and parameter, with dual-function knobs for gain/reverb, bass/compressor, mid/noise gate, treble/presence and volume/depth. The Preset encoder can be held down to save patches, while the Model one can be held down to access the setup menu, and is also used to change amp and cab selections. Holding down Parameter for a few seconds switches to an Alt mode, to access secondary functions for the rotary controls, whereas pressing it briefly accesses an advanced menu that allows you to adjust things such as the compressor and gate settings, the EQ frequencies, the reverb type and settings, and to toggle the EQ pre/post the amp model. You can also adjust the overall level of the amp model volume here. The three footswitches, which have all have status LEDs above them, can be used in pairs to change banks or individually to select a preset from the current bank. They can also be used to bypass the pedal.

On the rear of the pedal are quarter-inch jack connections for a mono input, stereo outputs, headphones and an external controller. MIDI in and out are on conventional 5-pin DIN sockets, and there's also the aforementioned USB port. The power connector is the familiar barrel type and a suitable 9V, centre-negative PSU is included.

In addition to Tonex amp capture, the pedal also supports IK's VIR multi-IR cabinets, and Tonex also has a custom IR Loader for adding user cabinet IRs. There's also an onboard noise gate, EQ, a compressor and five stereo reverbs taken from the X-Space pedal. There's comprehensive MIDI control support for those who need it, and an expression pedal input that can be assigned to any parameter. With both MIDI in and out, integration with other pedals or a MIDI-controlled pedalboard is possible.

In Use

Since the Tonex Max software can run both standalone or as a plug-in within your DAW or AmpliTube 5, you get a lot to play with even before you plug in the pedal — indeed »

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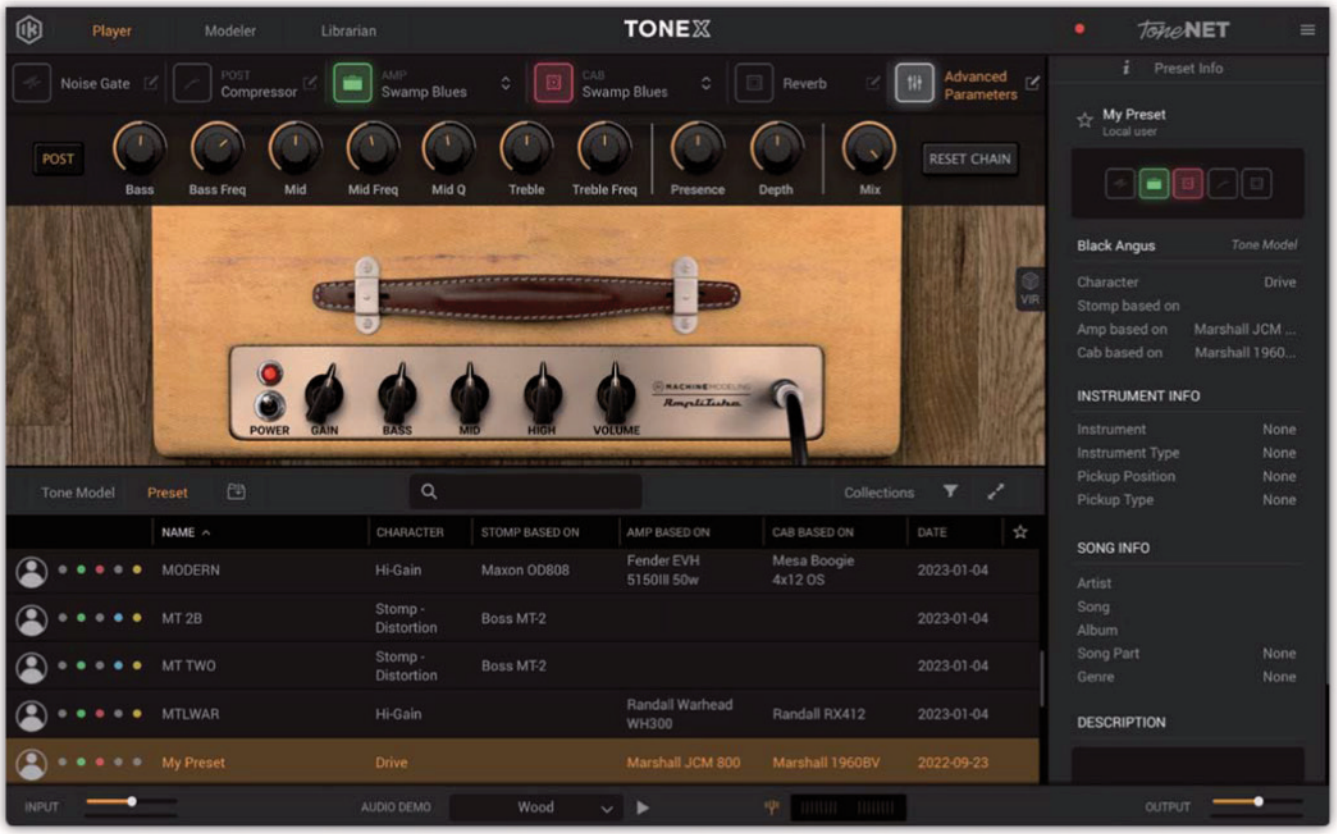

The Tonex software allows you to capture the sound and feel of real amps, and transfer them (or pre-captured models) to the Tonex pedal.

» you can demo the software for free before buying the pedal if you wish. And as the Tonex Pedal doubles as an audio interface, you can get a signal into your DAW without having to go back into the analogue domain. A USB connection is needed to communicate with the Tonex software, and if a firmware update is needed (as it was in my case), the software detects this automatically and takes you through the update.

Tonex's Librarian section encourages the browsing, loading and editing of Tone Models and presets, using an intuitive drag and drop interface to copy Tone Models from the library into any slot in your pedal, at which point you get asked if you want to overwrite the existing patch. The pedal syncs with the software when connected, and the Librarian shows your pedal presets at the top of the page and the available Tone Models at the bottom. Any user setting adjustments that you make in the Librarian can be loaded into your pedal, but often it's just as easy to load an existing Tone Model, then use the controls on the pedal itself to fine-tune the EQ, compression, reverb and so on before re-saving it.

I compared presets hosted in the Tonex software with the same preset running in the pedal and couldn't hear any difference. As the pre-loaded presets are a very mixed bag covering many styles, some will appeal to you and others won't. It makes sense, then, to organise all the presets you like at the start of the list, which you can do easily in the Librarian by dragging, at which point you are given the option to replace or swap with the existing patch.

What's really great about Tonex is that it's not just about the tone, but the playing feel of the amp is also captured, along with a real sense of the low-end weight that in my experience is so often lacking in amp modellers. In fact, the only omission I can see — and that I pointed out in the original Tonex software review — is that there's no way to add a tremolo to any of the amps. I'd have thought that should be possible without overtaxing the DSP.

The USB port is used for connection to a computer, both for communication with the Tonex software and to allow the pedal to act as a USB audio interface.

Verdict

In summary, then, the pedal is a very welcome addition to the Tonex family. It is easy to navigate, and it holds more presets than anyone could reasonably expect to use during a gig or studio session. The models sound great, and the playing feel is captured. And given that you also get the Tonex Max and AmpliTube 5 software included, the price is more than reasonable.

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Voyage Audio Spatial Mic Dante

Ambisonic Microphone

Has the addition of Dante allowed Voyage Audio's Ambisonic mic to realise its full potential?

SAM INGLIS

There are times when an otherwise great product is held back by a single limitation. Voyage Audio's Spatial Mic, which I reviewed in *SOS* October 2020 (www.soundonsound.com/reviews/voyage-audio-spatial-mic), is a good example. This is a nice-sounding second-order Ambisonic mic at a price to rival most first-order mics, and as it has built-in preamplification and A-D conversion, it does not require expensive matched preamps and converters. However, in a music-recording context this second positive is also a negative. The Spatial Mic can connect either to a host computer over USB, or to an audio interface over ADAT Lightpipe. Both options mean it can only be used with extremely short cables, and make it very difficult to record with other microphones simultaneously.

I ended my review of the original Spatial Mic by wondering whether Voyage Audio might consider developing "a 'pro' version that could be powered and connected to an audio-over-IP network using an Ethernet cable". And evidently I wasn't the only person who felt that way, because that's exactly what they've done. The Spatial Mic Dante is, well, a Spatial Mic that hooks up to a Dante network over Ethernet, presenting up to eight channels of audio at sample rates up to 192kHz. This is already an improvement over its predecessor, which topped out at 96kHz over USB and 48kHz over ADAT Lightpipe. A 48kHz-only AES67 mode is also supported for connection to other audio-over-IP networks, though I did not test this.

Inferno

The Spatial Mic Dante is similar in shape to the original but marginally larger, with a tough but low-key matte black finish. This time around, however, there's no headphone socket and no physical gain control. The assumption is that anyone using a Dante network will have a proper monitoring setup and access to a computer to adjust Spatial Mic parameters, which seems entirely reasonable. If the Dante version feels more substantial than the USB/ADAT version, that's mainly down to the presence of a large metal heat sink, which is clearly needed, as the mic gets quite warm in use.

Mounting arrangements are carried over from the USB/ADAT version, meaning that the mic itself has a 1/4-20 socket of the type used on camera tripods, which can be paired with the supplied ball-and-socket adaptor if you want to mate it to a mic stand. Given that this is a much more expensive and pro-oriented product than the original, it might have been nice if Voyage Audio had included the optional Rycote InVision mount and a proper carry case.

The base of the mic houses a Neutrik NE8FBH-C5 locking etherCON connector, which is also compatible with standard »

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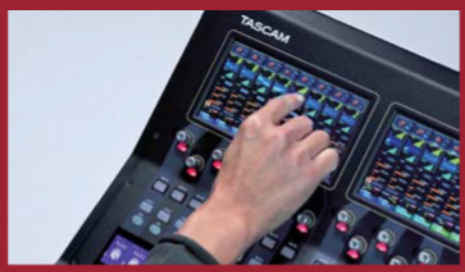

MAIN FEATURES

- Multiple 7" inch touch panels (3: Sonicview 24XP; 2: Sonicview 16XP)
- Integrated 32-track multi-track recorder (US version)
- 32in/32out, 32bit/96kHz audio interface function
- 96kHz, 54bit float FPGA mixing engine
- 32 bit/96 kHz AD converters
- Class 1 HDIA mic preamps
- Built-in 64in/64out Dante interface
- 44 input channels (40 mono inputs, 2 stereo inputs)
- 4 multi-effect processors
- Control software: TASCAM Sonicview Control supports Windows/MacOS/iPadOS for remote control and offline editing

Optonal stage box

- SB-16D 16in/16out Dante Stagebox

TASCAM

Voyage Audio Spatial Mic Dante

$2899

PROS

- The Dante format is perfect for the job, allowing the mic to be connected alongside other devices, by a single cable of practically unlimited length.
- Built-in processing allows a usable signal to be derived without plug-ins.
- Offers significantly better directional resolution than a first-order Ambisonic mic.

CONS

- Additional hardware needed to deliver Power over Ethernet (PoE).
- At the price, it would be nice to have a shockmount and hard case included.
- There are hoops to jump through with Logic Pro X at present.

SUMMARY

Voyage Audio have delivered a 'professional' version of their Spatial Mic that is an excellent showcase for the advantages of audio over IP.

» RJ45 plugs. Either side of this are two LEDs indicating the presence of power and network activity.

Purgatory

The Spatial Mic Dante is not quite a 'plug and play' device, and there are a few steps you need to take in order to get it working. In order to do anything at all with the mic, you'll need to be able to deliver up to 4.8W to it using the Power over Ethernet (PoE) protocol. And whereas power delivery is an integral part of the USB and Thunderbolt protocols, it's very much an optional extension of Ethernet. Many computers have Gigabit Ethernet sockets built in, and there are many USB and Thunderbolt adaptors that can add it to a machine that doesn't; but you won't get PoE support without an additional powered switch or PoE injector. These are readily available and affordable, and there are even battery-powered models available, but take care to get one that fully supports the IEEE 802.3af specification, as the Spatial Mic Dante requires 48V rather than the 24V that some injectors deliver.

Naturally, you'll also need cables, and bog-standard Cat5 worked fine in my tests. One of the biggest advantages of the Dante version over the original Spatial Mic is that Ethernet supports enormously long cable runs, and anything that works for other Dante devices should be fine with the Spatial Mic Dante.

Finally, you'll need to install some software. The Spatial Mic Dante has its own 'control panel' application called MicNetControl (see box). This, in a sense, makes the Spatial Mic Converter plug-in somewhat optional, though I imagine all users will want to install that too. You'll also need to visit the Audinate website to install the Dante Controller software; and, if the Spatial Mic Dante is your only Dante device, you'll also need a licence for the Dante Virtual Soundcard, which starts at $49. Once all this is in place, you simply tick a few boxes within Dante Controller and select DVS as your input device, if necessary, whereupon you should see signal appearing in your DAW.

As described in the 'MicNetControl' box, the signal-processing capabilities built into the Spatial Mic Dante permit a usable output to be derived directly from the mic. Where possible, though, it's better to record the raw eight-channel output and use the Spatial Mic Converter plug-in to derive your chosen signal, not least because that seems to be the only way to take advantage of the additional directivity that second-order Ambisonics makes possible. The internal DSP can generate virtual mics on the usual omni-to-figure-8 spectrum, as is possible in first-order Ambisonic mics, but if you want second-order cardioid, MaxRe or In »

MicNetControl

A second-order Ambisonic mic like the Spatial Mic has a coincident array of eight capsules. However, the raw output from these capsules is not in itself an Ambisonic signal but a precursor, usually referred to as A-format. In order to generate a second-order (nine-channel) or first-order (four-channel) Ambisonic signal in the B-format, these must be matrixed and filtered. Further processing is then required either to synthesize virtual microphones from the B-format signal, or to map it onto a particular output format such as a 5.1 or 7.1 speaker array.

The original USB/ADAT Spatial Mic can only output the raw eight-channel signal from the capsules, meaning that downstream processing is required to convert this to B-format and then to a suitable output format. That's what the Voyage Audio Spatial Mic Converter plug-in does, and this will usually be the preferred option, but there are some circumstances where you might want to do things differently. For example, you may need to use the Spatial Mic Dante with a DAW that doesn't support multi-channel tracks, or you might want to route its output directly to another node in the Dante network without passing through the plug-in, or perhaps you just want to record a simple mono or stereo signal without wasting drive space.

Enter the Spatial Mic Dante's built-in DSP, which can perform some of the same functions as are available in Spatial Mic Converter. This is set up using a standalone application called MicNetControl, which offers four processing modes. The simplest and default option is Unprocessed, which passes the raw eight-channel signal to your DAW for matrixing and decoding later. In this mode, the only controls available are a Mute button and a simple Gain slider with a 35dB range. This is the only mode available at sample rates above 48kHz.

In AmbiX mode, the individual signals from the eight capsules are matrixed and filtered to generate a four-channel, first-order Ambisonic output. Surround mode, meanwhile, matrixes the raw signal to Ambisonics before mapping it onto a 5.1, 5.1.7 or 7.1 speaker array. Finally, Pattern mode synthesizes a stereo virtual microphone array of your choice.

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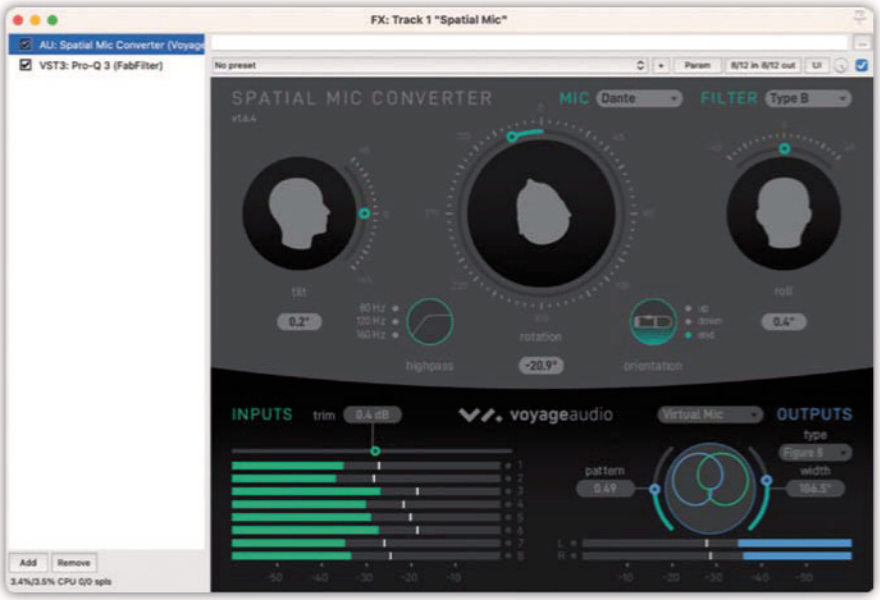

The Spatial Mic Converter Audio Units plug-in running in Reaper.

» Phase outputs you'll need to record the raw output and use the plug-in to derive these (see my review of the original Spatial Mic for more detail). Likewise, the internal DSP can generate a first-order B-format output, but not a second-order one.

Unlike the internal DSP, Spatial Mic Converter also offers a choice of two filter sets to be used in processing the eight-channel raw signal. Voyage Audio say that "Type A filtering exhibits high spatial resolution, while Type B offers a more diffuse, flat response." They sound surprisingly different from one another, with Type A being considerably brighter and more inclined to emphasise transients.

Paradise?

I wasn't able to compare the Dante version of the Spatial Mic to the original, but it certainly sounded good to me. Self-noise is theoretically a concern with a mic that derives its output from eight small-diaphragm capsules, but the review model was whisper-quiet, and the gain range of 35dB seems adequate to cope with most real-world recording scenarios. I should point out that although the Spatial Mic Dante can generate a 32-bit floating-point output, this doesn't offer any protection from clipping, as it does in the Rode NT1 5th Gen and other devices that use multiple converters in parallel. You might as well record the 24-bit fixed-point output and save yourself one byte in every four.

For the sake of variety, this time around I did my testing in Reaper and Logic Pro X rather than Pro Tools, and this unearthed

The Spatial Mic Dante uses a locking Neutrik connector that will also accept standard RJ45 plugs.

a few quirks. When you insert the AAX version of Spatial Mic Converter in a Pro Tools mixer channel, you have to choose an output channel configuration. If you were deriving a stereo virtual mic array, for example, you'd choose the 7.1-to-stereo version of the plug-in, and every subsequent plug-in on that channel would see a stereo signal. In AU and VST hosts, by contrast, there's just one version of Spatial Mic Converter and it always has a multi-channel output. If you use it in Virtual Mic mode to synthesize a stereo array, the L and R channels within the surround output carry signal, but the rest are still present, albeit silent. If you want to downmix the channel back to stereo you'll need to use something else to do it.

That's all perfectly manageable within Reaper, but Logic throws some additional spanners into the works, because of Apple's predilection for mucking about with the order of the channels within a surround signal. In order to get sensible results from Logic, you need not only to select the correct surround format for the Project and your input, but also to adjust the default input ordering in Settings / Audio / I/O Assignments. Voyage Audio provide a helpful guide on their website (https://voyage.audio/getting-started-with-logic-pro-and-spatial-mic/) but it might be better to simply include a Logic mode in the Spatial Audio Converter plug-in that remaps things automatically.

Talking of which, I'm told that other plug-in improvements are in the works already, including visualisation that will reveal where incident sound is coming from. This is already a feature of Rode's Soundfield By Rode plug-in, and hopefully should be even more useful in a second-order Ambisonic mic with its superior ability to resolve directional information. Likewise, MicNetControl should enable firmware updates that can improve the built-in processing still further.

In short, the Spatial Mic Dante is exactly what I and others have been asking for from a 'pro' version of the Spatial Mic. It offers the same, uniquely flexible virtual microphone functionality without being tethered to your laptop by a 2m cable. It can freely be combined with additional Spatial Mics and any other Dante-enabled hardware in a configuration of your choosing. And although it's more expensive than analogue second-order Ambisonic mics, it doesn't require multicore cables or matched preamps, has built-in processing, supports cable runs that are practically unlimited, and can be configured with one click from the comfort of your keyboard. Voyage Audio have hit the nail squarely on the head.

ALTERNATIVES

If you're looking specifically for a second-order Ambisonic microphone, the main alternatives are the **Core Sound Tetramic** and the **Brahma Studio 8**, while first-order options include the excellent **Soundfield By Rode NT-SF1** and the **Sennheiser Ambeo**. However, all of these are purely analogue devices, so as well as matrixing and decoding plug-ins, they also require matched preamps and A-D converters.

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The Osmose is unique among MPE instruments in that it offers the familiarity of a traditional keyboard layout — indeed, you can play it just like a normal MIDI keyboard, potentially making it the only controller you'll ever need. Don't be fooled by its looks, though: those keys are enormously sensitive, and respond to a huge range of expressive gestures such as striking, movement along both the X and Y axes for modulation, and, of course pressure: you can even 'squeeze' notes into existence from nothing with just a gentle touch. When we reviewed the Osmose back in *SOS* April 2023, we were glowing in our praise of the keyboard's action: "It's a truly exquisite level of sensitivity on offer, with a feeling of in-the-moment immediacy that can be breathtaking. Yes, it's a simple synth-like keyboard, in one sense. But this action also has aspects of response that are closer to various percussive instruments, perhaps hand drums or metal pans. Or, just as often, it gives the sensation that your finger has become bow- or breath-like."

But the Osmose is more than just a controller: embedded within it is the EaganMatrix, a powerful polyphonic synth engine designed specifically for MPE. Boasting 24-note polyphony, it incorporates virtual analogue, FM, physical modelling and granular synthesis. With five oscillators, plus noise sources, five multi-mode filters, built-in effects and even a convolution module, the EaganMatrix is capable of generating a huge range of timbres, from acoustic-like plucks to FM percussion, and everything in between.

For this month's competition, we're giving you the chance to win an Osmose of your very own — plus the complete Expressive E Instruments Collection, a bundle of five MPE-ready software instruments worth an additional $435! To enter, simply follow the URL shown and answer the questions there, by Friday 7th July 2023. Good luck!

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Neumann KH120 II

Active Monitors

Neumann continue to set new standards in compact nearfield monitoring.

PHIL WARD

If you read my review in the January 2023 issue of the Neumann KH150 you'll know that I was more than just a little impressed. To my ears, the KH150 significantly raises the standard of what's possible from a reasonably compact and relatively affordable active nearfield monitor. And now, in what seems almost indecent haste, Neumann have taken the engineering philosophies and electronic innards of the KH150 and integrated them in an updated version of their smaller KH120 model. So, here's the KH120 II. You may wish to refer to the longer and more comprehensive KH150 review for a little more background: https://sosm.ag/KH150.

The original KH120, launched in 2010, is Neumann's longest-established monitor and was the first new model to be introduced following parent company Sennheiser absorbing the venerable Klein + Hummel speaker company in 2005. The KH120 was, in fact, closely related to the Klein + Hummel 0110 model that was reviewed in these pages back in 2007. Of course the KH designation of all Neumann's monitor model numbers is a respectful nod to Klein + Hummel — founded, would you believe, way back in 1945.

Spot The Difference

Although slightly larger than the original KH120, the KH120 II is significantly more compact than the KH150. And rather than incorporating a 165mm (6.5-inch) bass/mid driver, it makes do with a smaller 130mm (5-inch) unit. The 25mm-dome tweeter fitted to both models is the same, but the elliptical waveguide of the KH120 II is necessarily slightly smaller than that of the KH150. A significant difference between the KH120 II and its predecessor is that where the enclosure of the original model was a fully die-cast aluminium component, the new model is constructed using the same technique as the KH150, with a wooden carcass and composite polycarbonate front and rear mouldings. That might seem like a downgrade, but Neumann say that the new model's enclosure construction technique actually results in improved resonance performance (very rigid, die-cast aluminium enclosures can be prone to resonant ringing, much like bells are), while also offering dramatically better energy efficiency in manufacture. Around the back of the enclosure are a couple of M6 threaded inserts that enable the monitors to be installed using one of Neumann's wide range of mounting accessories.

In terms of aesthetic style, the KH120 II follows the established Neumann monitor design language very closely, and without size reference in photos it's not easy to tell it apart from the KH150. The Neumann style isn't going to win any beauty prizes (to my jaded eyes anyway) but it looks suitably serious and professional, and is finished to a high standard. And speaking of finish, the review pair of monitors were dressed in Neumann's corporate, charcoal-coloured textured paint, but there's a white option available that looks classier to me. That's probably what I'd go for if I was buying my own pair. And while I'm on the subject of options, the KH120 II is available in both a standard version that offers just conventional analogue and digital audio inputs, and a network-enabled version that also makes possible connection to »

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Driving Force

The bass/mid driver is a new design and Neumann say that it follows the same principles as the driver fitted to the KH150, which emphasise linearity and low distortion — especially at high sound pressure levels. This is significant to my mind because I believe it's partly the implementation of those principles on the KH150 that makes it such an exceptional performer. Like the KH150, the KH120 II employs DSP-based bass/mid diaphragm excursion limiting (at a notably generous ±10mm) that helps improve the driver's inherent linearity. This is because DSP limiting means the driver suspension does not require the traditional in-built mechanical soft limiting that would otherwise be necessary but which has the down side of, in effect, introducing non-linearity.

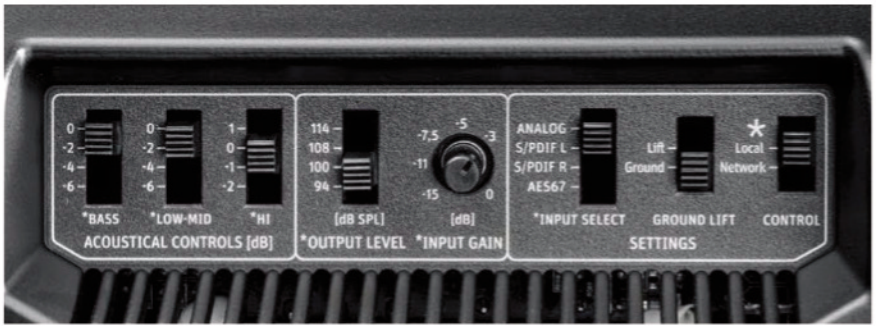
The KH120 II bass/mid driver is reflex loaded by two triangular cross-section ports that exit at the lower corners of the front panel. Again, as with the KH150, the KH120 II ports are flared both externally and internally and designed to remain linear to as high a volume level as possible while also not encouraging the emergence of organ-pipe resonances. The KH120 II's bass/mid driver and its reflex-loaded enclosure, aided of course by some DSP massaging, yield some notable specifications. The low-frequency cutoff (-3dB) is at 44Hz and the max sound pressure level (half space, 100Hz to 6kHz, <3% distortion) is 117dB. These are pretty extraordinary numbers for a compact monitor, and by way of comparison, the equivalent benchmarks for the original KH120 (which was no slouch) and the KH150 are 52Hz and 111dB, and 39Hz and 119dB, respectively. So in terms of low-frequency bandwidth and maximum level the KH120 II is rather closer to the KH150 than it is to its predecessor.

Neumann KH120 II
From $1998

PROS
- Fabulous clarity, imaging and neutrality.
- Great bass extension and volume capability for the size.

CONS
- None.

SUMMARY
With the KH120 II Neumann have pulled off much the same trick they managed with the KH150. It's a stupendously capable compact nearfield monitor and a more than a worthy successor to the original KH120.



The rear panel hosts a number of switches for adjusting the input gain and output level, tailoring the speaker's frequency response, and selecting between local and network control modes.

The bass/mid driver hands over to the tweeter at a conspicuously low 1.7kHz, and that's partly made possible by the mathematically modelled waveguide loading of the tweeter. The term 'loading' is used because the waveguide modifies the acoustic impedance (the acoustic 'load') of the air that the tweeter dome radiates into, and that results in increased radiation efficiency. The increased efficiency of the tweeter then means its usable bandwidth can extend down to a lower frequency than it would without a waveguide. And in addition to increasing the tweeters efficiency, the waveguide also serves to modify the tweeter's directivity so that it better integrates with that of the bass/mid driver in the crossover region. The tweeter is the same 25mm-diameter dome unit used in the KH150 and other Neumann monitors.

Switching It Up

Amplification comprises the same 145W and 100W amplifiers as fitted to the KH150 for the bass/mid and tweeter elements respectively. Also shared with the KH150 is the KH120 II DSP engine and its extensive configuration functionality. That functionality is manifested on the KH120 II's rear panel by a range of EQ facilities that enable bass cut to -6dB in 2dB steps, low midrange (around 300Hz) cut to -6dB in 2dB steps and higher-frequency adjustment from +1dB to -2dB in 1dB steps. There's also both input gain and input sensitivity (expressed in terms of acoustic output level for 0dBu input) controls, and an input selector switch that offers analogue, S/PDIF (left), S/PDIF (right) and S/PDIF (mono) options, and a ground-lift switch.

Alongside the ground lift is a further switch labelled 'local' and 'network'. In the local position, a KH120 II is configured in EQ terms by its rear-panel switches, while in the network position it is controlled by Neumann's MA-1 room optimisation and monitor configuration app. Said integration is achieved via Ethernet sockets on the rear of the KH120 II that enable them to be connected to the same TCP/IP network as the computer running the MA-1 app.

MA-1 conceptually follows the pattern set by other similar systems such as Sonarworks SoundID, Dirac Live, IK Multimedia ARC, Trinnov and Genelec's GLM. A calibrated measuring microphone is used to take in-room acoustic data created by routing a predefined swept sine-wave signal to each monitor in turn. The mic output data is then used, via some sophisticated interpretative algorithms, to create a response profile for each monitor that pre-equalises their outputs to compensate for the influence of the room acoustics. The MA-1 configuration process requires measurement data to be taken in seven measuring positions dotted around a primary listening position, and once that's done the MA-1 app takes just a few moments to generate the appropriate equalisation profile for each monitor. I've included a couple of MA-1 screenshots that illustrate the before and after response, as captured by the MA-1 app of the pair of KH120 IIs in my studio room. Integration of the DSP-driven MA-1 technology of course means that the KH120 II works entirely in the digital domain internally. As with the KH150, the internal sampling rate is a modest 48kHz, fixed, say Neumann, to ensure that the tweeter is not driven much above »

ALTERNATIVES

The KH120 II is hugely capable but it's not alone. Options such as the **Genelec 8030A**, **HEDD 07 Mk2**, **IK Multimedia iLoud Precision 5**, **Focal Solo6 ST6**, **Dynaudio LYD 8** and **ADAM A7V** might all be worth considering.

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The in-room response of a pair of Neumann KH120 IIs, before and after MA-1 optimisation.

» 20kHz because doing so would risk dome break-up distortion products folding down into the audible band.

Measuring Up

I took one KH120 II to my speaker measuring space and launched FuzzMeasure to investigate a couple of aspects of its acoustic performance. Firstly, Diagram 1 shows the KH120 II axial amplitude frequency response at 1m distance. While not quite displaying the almost ruler-like linearity of the KH150, the KH120 II delivers an impressively flat line. Such measurements are finely sensitive to exact mic position, though, and if I'd spent some time experimenting with moving the measuring mic a few tens of millimetres in, out or sideways I could very probably have found a flatter sweet spot. However, the real significance of frequency response curves is found in general trends or specific discontinuities that might point to particular flaws. And in the case of the KH120 II the general trend is flat, and specific discontinuities are noticeably absent.

Diagram 1 also displays second- and third-harmonic distortion, and as with the KH150, both are extremely well controlled. The acoustic output level was 90dB at 1m and the distortion curves are generally around 50dB below the fundamental. Expressed in percentage terms this equates to around 0.3%. It's a very good result.

I mentioned earlier that the port design is specifically optimised to suppress the generation of organ-pipe resonance, and Diagram 2 illustrates Neumann's success. A port organ-pipe resonance can occur, as the name so clearly suggests, when a port tube displays a resonance primarily defined by its length. And due to typical port lengths, organ pipe resonances typically occur in the lower midrange, say 400Hz to 800Hz, where the human ear is inconveniently extremely sensitive, so they can be easily audible. Diagram 2 shows a frequency response captured with the measuring microphone located right at the mouth of a KH120 II port. Organ pipe resonances would normally reveal themselves through high-Q peaks in the curve as it decays above the port Helmholtz resonance (45Hz, in the case of the KH120 II). The KH120 II curve is completely clean until the port output decays into the noise floor. There's no sign of even the slightest resonance. This is truly unusual and genuinely impressive.

Listening In

So, describing and measuring done, I'll move on to listening. I've heard the original KH120 on a few occasions but never over an extended period in my own space, but I do still have the KH150 review samples in my possession, and the 'family likeness' between them and the KH120 IIs is unmistakeable. The KH120 II is in many ways equally impressive as the KH150. The two are clearly cut from the same cloth and driven by the same design

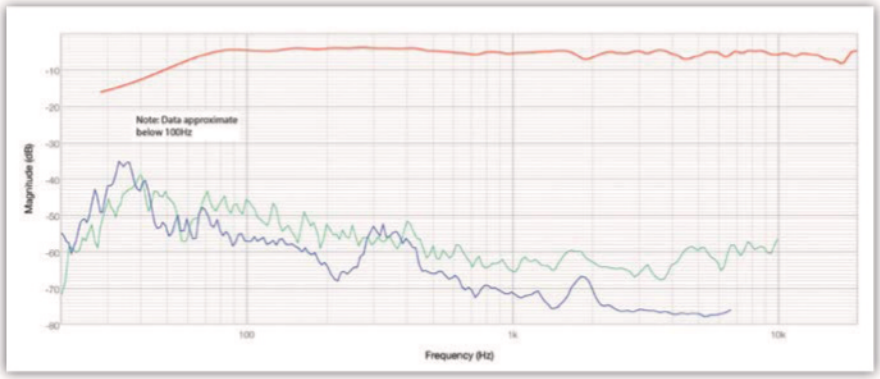

Diagram 1: An on-axis measurement of the KH120 II (red trace), and second- and third-harmonic distortion levels (green and blue, respectively).

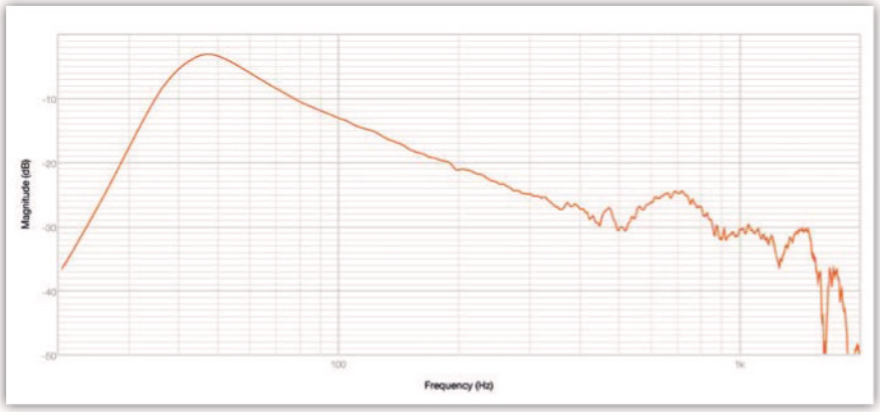

Diagram 2: A close-mic measurement of the reflex port output.

philosophies. This is especially the case with the MA-1 room optimisation working; in these circumstances they sound even more alike. And speaking of MA-1, my experience of it with the KH120 II was just as positive as it was with the KH150. In a reasonably well-behaved studio room, MA-1 doesn't transform the monitor's performance, but it definitely adds a confidence-building safety net. If I were a KH120 II user, I'd not hesitate to go with the MA-1 option.

Again like the KH150, there's a deep clarity and accuracy to the sound of the KH120 II. Its overall tonal balance is consistently and reliably neutral and its ability to create extraordinarily well-focused stereo images speaks of really well sorted electro-acoustic design. But it's not just tonal balance and imaging that impresses. The KH120 II reveals mix detail and insight throughout the audio band in a way that makes mix work a pleasure. There's no guesswork. It's all there. The slight reduction of low-frequency bandwidth and maximum level compared to the KH150 is noticeable in a direct comparison, but heard on its own, KH120 II bass is both impressively extended and usefully informative of pitch and dynamics. And I can't really imagine running out of nearfield volume level in any reasonable work scenario with the KH120 II. There wouldn't be too many occasions in a smallish studio space where you'd need a nearfield monitor with more bass and volume potential.

The KH150 and now the KH120 II demonstrate without doubt that Neumann's monitor design team not only have a very clear idea of what's needed in electro-acoustic terms to create really effective nearfield monitors, but how to put those ideas affordably into practice. They're on fire.

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Tascam Portacapture X6

Portable Multitrack Recorder

The Portacapture X6 makes Tascam's touchscreen approach to mobile recording more portable and affordable than ever.

CHRIS TIMSON

I had the privilege of reviewing Tascam's Portacapture X8 a year ago (*SOS* May 2022) and thought it a fine piece of kit — I was sufficiently impressed to part with my own money and buy one. Consequently, I was very willing to give its newer, smaller stablemate the X6 a try when offered the chance. It's very tempting to review a device like this in terms of its differences from the earlier model, but that would be doing the X6 a disservice so, instead, I shall consider the X6 on its own terms and then discuss the differences later.

Overview

What we have here is a portable recorder that can record up to four separate tracks simultaneously, plus a stereo mix track based on the first four (hence the '6' in the name). That's two from the in-built mics and two from the balanced XLR inputs on the left-hand side. Like the X8, the X6 is capable of 32-bit floating-point A-D conversion and recording, the chief advantage of which is that even if you pay no attention to the preamp gain you can capture a clean signal, whose level can be adjusted in post-production.

At the top of the recorder, the in-built mics provide inputs 1 and 2. These are not detachable but they can be rotated to form either X-Y or A-B stereo arrays. Directly below, on the front panel, is a very small speaker. As ever, no high-fidelity prizes for this but it's a useful tool to confirm that what you recorded is what you thought you recorded. Below that is the 2.4-inch colour touchscreen that, as with its big brother, lies at the heart of this machine's functionality.

At the bottom of the front panel, below the screen, is a set of transport buttons, and the stop control doubles up as a Home button for the software GUI. A thumb wheel is used for fine adjustment of input levels and virtual faders, and a Mark button sets markers on the recording, to facilitate fast navigation. There's also a useful menu button that I wish had been included on the X8 as well (oops... I said I was going to avoid comparisons!).

The left-hand side of the recorder has inputs 3 and 4, the aforementioned XLR microphone sockets. These are, of course, capable of supplying 48V phantom power. Next to those are two 3.5mm TRS mini-jack sockets. The first provides two channels of line input as an alternative to the XLR inputs 3 or 4, or it can be configured for the connection of a mono or stereo mic using plug-in power, again as an alternative to the XLR inputs, and useful for lavalier mics and the like. The second provides a stereo line output to other equipment. Together, these two sockets can provide connectivity to a video camera if required, and there are configuration settings to support this usage. Next there's a headphone socket and, finally, a pair of plus/minus buttons

Tascam Portacapture X6

$369

PROS

- Really easy to use.
- App-style interface caters for various uses.
- Optional Bluetooth adaptor and Android/iOS app.
- Good onboard mics and preamps for external mics.
- 32-bit conversion and recording.
- User presets a promised future enhancement.
- Can be used as an audio interface.
- Good price.

CONS

- File management and naming a little basic.
- The smaller screen can mean precise finger work is needed.
- SD card, external PSU and Bluetooth adaptor aren't included.

SUMMARY

A great-sounding, easy-to-use portable recorder that supports external mics and multitrack recording, and doesn't cost a bomb.

that control the volume of the headphone socket and the internal speaker.

The right-hand side has a slot for the micro-SD card (not included; up to 512GB micro-SDXC models are supported). There's a USB-C port that both acts as an external power inlet (you'll need to provide your own PSU for that) and caters for file transfer to/from a computer. There's also the on/off/hold switch. The end just has a slot for the optional Bluetooth adaptor, and peak and recording indicators. Underneath, a quarter-inch threaded insert enables mounting on a camera tripod; if you'd prefer a mic stand then adaptors are widely available.

The Portacapture X6 comes with a decent, printed 'get you going' manual but if you want to really learn the ins and out of this machine you'll need to download the full reference manual from the Tascam website.

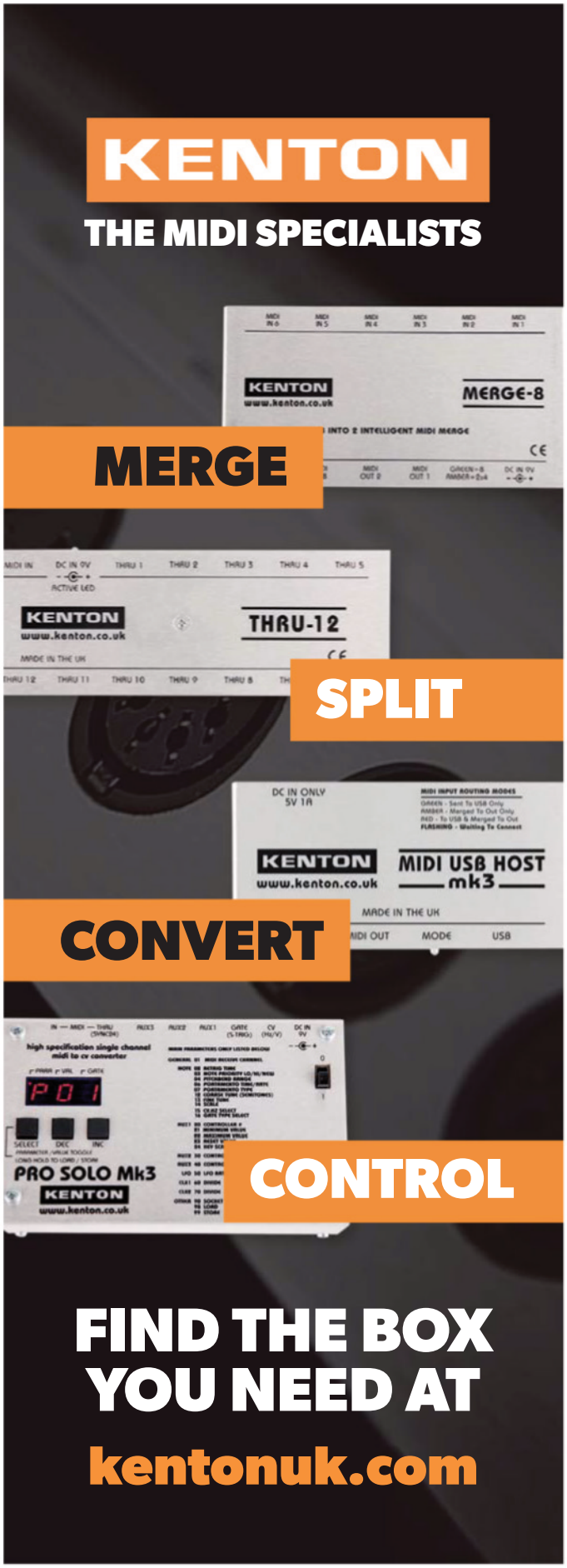
Launching Into It

Just like its older sibling, the main way you interact with the Portacapture X6 is less via the physical controls and more through the touchscreen, in an approach that owes an awful lot to smartphone design. This means that you can get quite a long way with this machine just by touching icons and seeing what happens. Just like a smartphone, you start with the Launcher, which presents you with a set of 'apps' that give you access to all the functionality of the machine. You get exactly the same set of apps in both the X6 and the X8, with the exception of two minor apps that are missing on the X6: a tuner and a metronome. In my review of the X8 I gave quite detailed descriptions of each of the apps and I commend that review to you for the full monty; for this review, I shall give just brief descriptions of each one.

Browse: I don't think I can be too brief with Browse, though, because to use it you have to understand the X6's filing system. Browse is where you can list and play projects, delete and rename projects, and obtain basic information about them. You can also create new folders here (to a depth of two) to contain projects. So, what's a project? It's all the files associated with a single recording. Multiple recordings are multiple projects, so if you want to keep all the recordings associated with a single real-world project, then you should create a folder to hold them. But you have to remember to change into that folder before you make your recording, and that's something that I kept forgetting to do! It's the same for the X8, and one of the few things I don't like so much — I'd much prefer something like the set of files for a single recording to be called something like a 'take' and a project to be defined as a collection of related takes.

All the files in a project share a common filename that's derived from a root name you can specify (otherwise it defaults to 'Tascam_'). There will be a mono audio file for each active mic, or a stereo file if two mics are stereo linked. There will also be a file for a mix of the active mics. New projects will take the current root filename and increment a number to identify it uniquely, eg. 'Tascam_0023'. For naming things there is a neat on-screen virtual keyboard.

ASMR: ASMR stands for Autonomous Sensory Meridian Response. This means it does pretty pictures while making »



The Portacapture X6 can capture two external mics, with the XLRs capable of delivering phantom power and the stereo mini-jack input capable of plug-in power.

» normal stereo recordings. Well, who'd have guessed that!

Voice: A quick and easy way of making voice recordings for interviews and the like. As an example of the use of the thumb wheel at the bottom of the recorder, you can tap on the gain display and adjust it with some precision by using the wheel. This behaviour is widespread through the GUI and always worth trying.

Music: For making stereo recordings of musical instruments. It provides a number of presets to adjust dynamics and suit different instruments (piano, guitar, stringed instrument and so on).

Manual: This one is, for me, the key app. It's where every setting can be tweaked and you have complete control. It uses a mixer metaphor and is pretty straightforward to use. Facilities include, among other useful things, a simple compressor and EQ. However, the smallness of the touchscreen and number of different parameters you can change using this app mean you have to be pretty precise with your finger... though I did find a very usable workaround via Bluetooth, of which more in the separate box.

Field: This is another stereo recording app that this time offers a set of presets appropriate to field recording tasks, such as City, Nature, Bird etc.

Podcast: Though obviously not the equal of dedicated podcast hardware, this app nevertheless offers a number of useful facilities to aid in the creation of podcasts.

General Settings: Here's where you set all the useful stuff like bit rate and dual recording.

Sounding It Out

All this is good stuff but not worth a fig if the recorder doesn't make decent recordings, since that is its entire *raison d'être*! In my review of the Portacapture X8, I set up a shootout between the X8 and a pair of Neumann KM184 mics, and was surprised and pleased at how close the quality of the X8 recordings came to the KM184s — it's one of the main reasons I bought my X8. For this review, I thought it would be interesting to compare the X6 with the X8 directly. My working assumption is that the two recorders use the same preamps, but it's much less certain that the mics are the same since the X6 mics are not detachable, and are also slightly smaller. I was keen to see if anything had been lost in the design changes.

As you can see in the photo, I mounted the X6 and X8 next to each other and set them recording using the Manual recording app, with similar settings in both cases. I recorded one instrumental tune (fiddle and concertina) and one unaccompanied vocal harmony song together with my wife Anne Gregson. I transferred them to my DAW (Logic Pro X) but, apart from normalising the volumes to make comparisons easier, I did no other processing on them. I then had a listen...

On the instrumental tracks, I couldn't hear any difference at all. In fact, I put

The mics on the X6 aren't identical to those on the X8, but testing the two recorders side by side on instruments and vocals revealed them to be equally impressive.

EQs on both tracks and watched the displays and could see little difference between them either. I thought I could hear a slight difference between the vocal takes — the X6 seemed a little weightier — but again the EQ's analyser showed very similar things happening. Either way, and in both cases, I would be happy to use the recording.

Sibling Rivalry

So, how do the X6 and X8 compare beyond the sound, which we've already established is comparable? I'll starting with the elephant in the room: size. When I reviewed the X8, I felt that as a handheld recorder it was approaching the limits of usability. With a set of batteries installed, mine weighs nearly half a kilo, and after a short while carrying it in your hand you definitely begin to feel it. The X6 weighs less (378g), and it's the same width as the X8 (so still fairly chunky) but I've found that the lower weight does make a real difference over time.

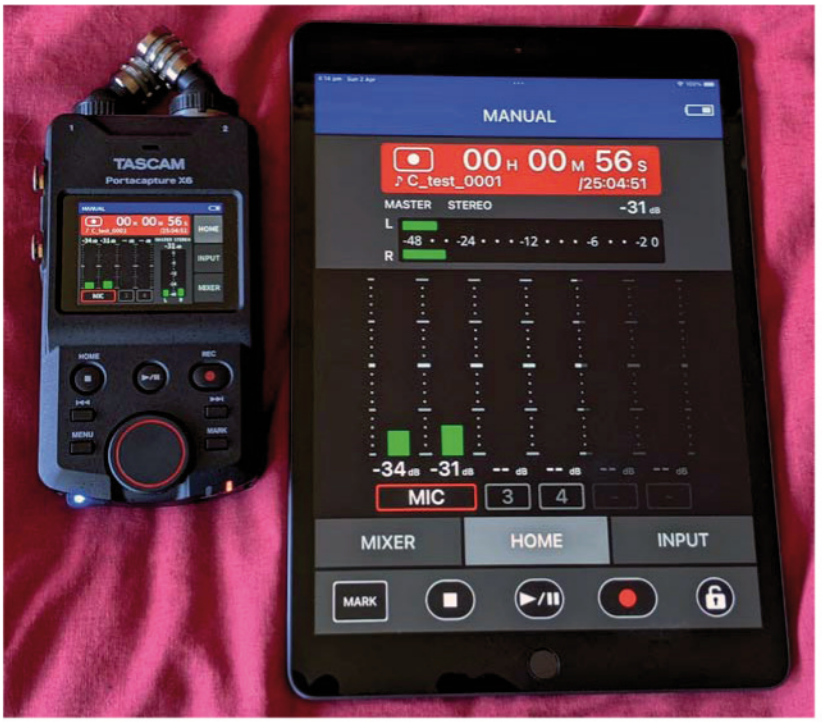
Of course the touchscreen is much smaller on the X6 than it is on the X8, but the designers have got round this in some clever ways — for instance the aforementioned X6 menu button, which replaces a menu icon on the X8's screen, thus freeing up a bit of screen space. They are also assisted by the fact that there are two fewer recording channels but they've shown a lot of ingenuity in remapping the screen objects in each app to fit the available space. In fact the only app where I felt significantly constrained by screen space was Manual.

ALTERNATIVES

Many of the interesting devices in this area seem to come from **Tascam** and **Zoom**. The **Zoom F3** offers many of the same facilities as the **X6** including 32-bit float and two high-quality XLR inputs with phantom power, but has no in-built mics, which means its use must always be premeditated. The **Zoom F6** is the **F3**'s big brother and shares its novel form factor but is more expensive than either the F3 or the X6. The **Zoom H6** offers many of the facilities of the X6 and X8 but lacks the ease of use of either and also the all-important 32-bit float recording. Personally, I think the biggest direct competitor to the Portacapture X6 is the **Portacapture X8**! The price difference isn't vast and the X8 offers two more inputs and a bigger screen — but it's also a bigger piece of kit to have to manage, and may have facilities you don't need. You pays yer money...

Sync Or Swim With Bluetooth

The X6 uses the same Bluetooth adaptor as the X8 (the AK-BT1). This costs about $39 and, in my opinion, if you have an Android or iOS device to act as the controller then it is an essential purchase. The Tascam Portacapture Control app, which I'm pleased to say runs on Android as well as iOS, gives you full control of both the X6 and the X8 using the same GUI, but laid out on a much larger screen — and remotely at that! This is particularly useful for the Manual app, as you can imagine. This app, which is free to download, is common to the X6 and X8, and those elements of the X8 GUI that aren't present on the X6 are greyed out.



The Bluetooth adaptor is available separately and allows communication with a free iOS/Android remote control app.

However I found a neat workaround for this via Bluetooth, as described in the box.

Finally, of course, the X8 has the two extra XLR inputs. Whether this is important to you only you can say, but I think this might well be the main deciding factor for many people who need to choose between these two recorders.

Future Features

Tascam say that there are a number of enhancements in the future for the X6, but the one I find most interesting is the idea of user presets. In fact, this is something I'm quite proud of because I suggested the idea to Tascam while reviewing the X8 and they took it up enthusiastically, implementing it in an update to the X8's firmware, and it works well. I'm pretty certain that the implementation will be much the same for the X6 as for the X8. The idea is that if you have a setup that you use frequently, a particular external microphone at a particular gain, or perhaps a specific setup for a podcast, then you should be able to save it for future use. This idea is commonplace in DAWs but I hadn't seen it in portable recorders before. You can save up to three user presets for inputs that are shared across recording applications, three for recording apps, and three for the whole unit.

The bottom line, though, is that Tascam have done it again. The Portacapture X6 is already a practical, easy-to-use field recorder and one that's capable of high-quality recordings. Like the X8 it works well, sounds good, is very easy to use, and is available for a very reasonable price. It deserves to sell well.

$ $369.
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W tascam.com/us

Erica Synths Perkons HD-01

Drum Machine

Perkons is the Latvian name for the Baltic god of thunder, a fitting name for a thunderous drum machine.

RORY DOW

Erica Synths are rapidly becoming one of the most prolific instrument manufacturers around. Just a year ago, I reviewed their excellent LXR-02 drum machine, and even then, they had already announced the Perkons HD-01, a chunky-looking drum and drone synth. Production has been slow thanks to global chip shortages, but now the Perkons is finally available *en masse*.

There are four drum voices, each with a choice of three synthesis algorithms. The oscillators are digital, whilst the multi-mode filters and overdrive circuits are analogue. The sequencer consists of four rows of 16 triggers, one for each voice, and many fun functions like probability, playback direction, modulation recording and trigger conditions. There is a global BBD-style digital delay, with separate send per voice and a single LFO that can modulate most voice parameters.

From a glance, it's easy to tell that the Perkons is designed for hands-on control. It follows the same desktop build format as the Syntrx, their EMS VCS 3-inspired semi-modular synth. It's a desktop unit, around 46 x 32 x 9cm. The wooden sides, aluminium powder-blue case and substantial knobs give it an air of class, and the sequencer section is populated with satisfyingly chunky buttons. It's an imposing-looking instrument, but do the sounds match the look?

Synthesis

Each digital engine has three algorithms to choose from, and each algorithm has three modes. Broadly speaking, voices 1 and 2 are more tonal — good for kicks, toms and synth sounds. Voice 3 covers snares and percussion, and voice 4 focuses on noise, hats and cymbals.

With each voice having three algorithms and three modes, there are 36 variations of synthesis algorithm to choose from. In reality, some modes only change minor details like a transient volume. In others, they are more like different algorithms, drastically changing the sound.

The panel layout for each voice is identical. Each voice, algorithm and mode offers the same pitch control, decay, filter cutoff, overdrive, effects send, overall level and two 'param' controls specific to the currently selected algorithm mode. This generic layout makes things simple. A knob will generally do the same thing no matter where you are. But the lack of algorithm-specific labelling means you are flying in the dark about what the 'param' controls are supposed to do. The algorithms and modes have numbers rather than useful names. Instead of memorising the algorithm and mode numbers or keeping the manual nearby, my preferred approach was to blindly switch modes until I found something I liked. Some extra information on the panel might have helped.

Voice 1 will likely be your kick-drum sound, although the algorithms allow you to stray far from that brief. You can choose between wave folding, wavetable and FM algorithms, with the three modes performing tasks such as transient shaping, wavetable selection, or in the case of the third algorithm, a more drastic tonal shift caused by changing the FM waveforms.

Voice 2 contains three flexible 'drum' engines. Like voice 1, they use wave folding, wavetable and FM synthesis, but the algorithms are tuned differently. This voice is more suited to percussion, synth, toms and effects.

Voice 3 will focus on snare, clap and tom-type sounds using noise, ratcheting, FM and Karplus-Strong synthesis. It introduces noise and would also suit more experimental sounds.

Voice 4 concentrates on hi-hat, metallic and noise-based sounds utilising different noise types and some specific closed, open and ride synthesis models.

Overall, the synthesis feels simple, bold and flexible. Despite its digital sources, the Perkons sounds massively analogue. The result is a sharp sound with plenty of body and enough low-end to knock a house down. That is assuming the neighbours don't call the noise pollution police first!

The excellent analogue multi-mode filters offer low-, band- and high-pass modes. No resonance control exists, but you will find a very usable analogue overdrive circuit. With no resonance, the filters are used more like an EQ to shape the oscillator sound. The only possible modulation is via the global LFO or by automating using the sequencer. It's a shame there isn't a way to use the amplitude envelope to vary the frequency, but that would have needed another control on the front panel.

One neat little bonus is the ability to turn the Perkons into a drone synth by increasing a voice's decay parameter to 100 percent. The VCA will stay permanently open, and the oscillator will drone indefinitely. Suddenly the Perkons is a very different synth, but the ability to automate the decay parameter means you can drop in and out of 'drone mode' with some simple automation.

Sequencer

Nowadays, a good drum machine can stand or fall by its sequencer, and the Perkons delivers plenty to get excited about. A pattern is limited to a maximum of 16 steps per voice, but you can chain multiple patterns in any order to create longer sequences.

The grid of 4x16 trigger buttons enables or disable steps in the sequence. There is also an optional Accent mode that allows two volume options per step. If programming patterns isn't for you, you can record live using the four trigger buttons, MIDI input or the analogue trigger inputs.

Tracks can play back in different directions — forward, reverse, ping-pong and random — with individual clock division, multiplication and loop points. Every step can also benefit from ratcheting up to five triggers within a single step. Per-step 'odds' also dictate how often a trigger will play. For example, set it to 1/4, and it will play every fourth pass through the sequence. Set it to 7/8, and it will play seven times out of eight. Finally, probability can be set to 10, 25, 50 or 90 percent, randomising the chance of a step triggering. These functions allow for much more complicated and evolving patterns than the initial 16 steps might allow.

The most potent trick up the sequencer's sleeve is the ability to store parameter changes per step. Any »

Erica Synths
Perkons HD-01
$2059

PROS
- Huge sound with bags of character.
- It's fun.
- Separate outputs and sends/returns for each voice.

CONS
- Only four voices.
- Completely mono.
- Quite expensive.

SUMMARY
The Perkons HD-01 is a unique drum machine with no real equals. It combines subtractive synthesis, FM, wavetables, cymbal synthesis and even Karplus-Strong with a fun techno-leaning sequencer that is fun to program and easy to tweak. Its hands-on interface and multiple audio sends and returns encourage live performances and sound design. All this whilst delivering a thunderous sound. Great fun.

Round The Back

All audio inputs and outputs are on quarter-inch TS connectors. Each voice has an output, trigger input, and send and return (allowing you to replace the master BBD delay per voice). There is also a master mix output, a headphone jack and a master send and return (replacing the BBD delay globally). Finally, there is an analogue clock input and output, MIDI in and out on 5-pin DIN, and an SD card slot for saving patterns, kits and firmware updates. Remember that the Perkons is a monophonic synth, so all audio outputs are mono. You can't use stereo effects on the send and return loops, and if you want stereo panning, you'll need an external mixer.

» combination of knob and switch automation can be stored, allowing for dramatic or subtle sound changes per step. Holding a step button and moving a control on the associated voice will commit that change to the step. You can even hold a modifier to store that parameter for all four voices at once, allowing for kit-wide automation. Alternatively, you can put the sequencer into record mode and, as it plays, move knobs and switches. The resulting parameter changes will be recorded in the sequencer. It is a step sequencer, so the recorded automation is quantised to the steps of the sequencer, but you can enable a glide function, which can smooth out the automation between steps. All these features can be used in tandem while the sequencer is running, making the Perkons a live performer's dream.

There are some additional tools to help in your programming adventures. For example, you can easily copy and paste individual steps or delete automation for the whole pattern or individual steps when things get too crazy.

Patterns can be saved, as can kits. By default, kits and patterns are saved and loaded separately, which means if you have a killer drum groove, you can try out other kits. But if preferred, you can enable 'kit linking', which will load the corresponding kit when loading a pattern.

The Perkons can save up to 64 patterns and kits in a bank, and the SD card can store up to 64 banks. That's a whopping total of 4096 kits and patterns. There is a slight pause when switching between banks while they load from the SD card, so if you need instant kit or pattern switching for a live performance, you must stay within a bank. I suspect 64 patterns will be plenty for a live performance though, so no problem there. Speaking of performances, anyone playing live will appreciate the many ways to sync the Perkons to other machines. The usual MIDI Clock options and analogue sync with adjustable ppqn are available. You can also tap tempo for manual sync, and tempo nudge for keeping time with other musicians or DJs.

The final sequencer features worth looking at are groove and shuffle. Groove allows you to choose one of four groove templates for each voice. These grooves apply a volume map, varying the volume on specific steps. There are three preset grooves and a random one. Shuffle uses a standard 16th swing. Applying shuffle and groove per voice makes for combinations that can subtly or drastically change the feel of a pattern.

L... F... O...

The Perkons' LFO is another fun source of modulation. It can be assigned to any of the eight knobs for each voice: tune, decay, param 1, param 2, filter frequency, drive, effects send and level. There are seven waveforms and a simple master speed and master level control. Although it's bare-bones, it helps create some non-sequencer-specific movement in a kit and can be used subtly or drastically as you see fit. The only thing I miss, something that the Erica Synths LXR-02 has, is an LFO reset feature. Being able to reset the LFO using a sequencer voice trigger was something I used a lot, and I miss it here. LFO setting are saved with the kit.

Master Section

As well as the overall volume control, the master section contains the controls for the BBD-style delay and the master compressor. The compressor is only audible when monitoring the main output, but the thoughtful inclusion of send and return jacks for each voice means that you can process each voice individually (bust those guitar pedals out!) and then loop the effected voices back into the machine to be processed by the delay and compressor. You can take this one stage further by using the master send and return jacks to replace the delay with some other effects and return the mixed signal for final compression.

The included delay is excellent. It has an analogue character, despite being completely digital. Alongside its time and feedback controls is a switch to alter the time range from short to long. With short times and the use of a second dedicated LFO, it's possible to do a variety of modulation effects and feedback tricks. The LFO is a simple sine wave with rate and depth, affecting only the delay time. You can conjure chorus, flanger-type effects and some wilder wobbles with it. Longer delay times allow for more traditional dub-style delays. Apart from another switch that changes the colour of the delay tail (dark or bright) and a hidden preference that purports to give a more vintage character (although I couldn't hear much difference), there's not much else to it. Nonetheless, it is a very useable effect and sounds so good you'll likely overuse it.

MIDI Control

We've already mentioned MIDI and analogue clock sync, but there is also an excellent MIDI implementation for those wanting to sequence the Perkons from a DAW or external sequencer.

The Perkons measures 315 x 454 x 90mm (including the knobs) and weighs in at a hefty 3.7kg.

Each voice can be triggered by analogue triggers, by different notes on the same MIDI channel, or by the same note on different MIDI channels. Sadly, you can't play voices chromatically, which would be a nice improvement in a future firmware update. One advantage of sequencing with MIDI is velocity sensitivity. You get full volume range via velocity instead of the on, off or accented choices when using the internal sequencer (although you can use per-step automation for level as a workaround).

Each voice parameter also has its own MIDI CC assignment, which can be sent and received. There doesn't seem to be any way to control the master delay, compressor or LFO, however, which seems a shame. If the default CC assignments don't suit you, you can edit a text file on the SD card to change them.

Conclusion

On paper, a four-voice drum machine doesn't seem all that appealing, especially at this price. But I guarantee that perception will change as soon as you hear it. Perkons lives up to its namesake, the Baltic god of thunder, with a colossal sound that doesn't fail to impress. Kicks range from tuned sub-bass heavyweights to spectrum-filling distorted gabber giants. Percussion is snappy and sharp, covering everything from analogue toms, ratchet claps, laser zaps, claves, blocks and Karplus-Strong plucks. Hi-hats and cymbals sizzle brightly, and although there are occasionally some piercing frequencies, you can almost always eliminate them with careful tweaking. Overall, the Perkons excels at big, brash sounds. It is capable of subtlety, but it can be harder to achieve. If you want 808 or CR-78 style finesse and soft edges, this maybe isn't the right drum machine. The Perkons is the Muhammad Ali of drum machines. It punches hard and with effortless grace.

The design is hands-on, and once the muscle memory kicks in, your fingers will be dancing around the front panel like Jeff Mills on a 909. The performance workflow of saving patterns, then slowly morphing them to something new before returning to base with a reload, is possible here. And with 64 instantly recallable patterns and kits, there is enough to keep the party going without ever stopping the transport.

The Perkons also encourages creativity. I love the audio send and returns. I spent a most enjoyable evening with the Perkons on the rug, a bunch of cables, and some effect pedals, experimenting with different chains on individual voices and the master channel.

There are some limitations, of course. If you're a fan of stereo drums, this may not be the drum machine for you. Alternatively, you might accept the trade-off and process a couple of voices externally with stereo effects. Also, it's unlikely to be your one-and-only drum machine. With only four voices, it does need some backup occasionally. However, those four voices are quite capable of filling the frequency spectrum. Less is more in this case.

Erica Synths are on a roll. They consistently bring out unique instruments that are fun to play. The Perkons HD-01 is no different. It has sonic character by the truckload. I think the Latvian god of thunder would approve.

$ $2059
W www.ericasynths.lv

Tierra Audio New Twenties

Cardioid Capacitor Microphone

This handmade mic offers vibe in abundance.

Tierra Audio
New Twenties
$1052

PROS
- Stylish retro design.
- Characterful sound — but not overbearing.
- Low-frequency roll-off works well for close miking.
- Frequency response works well for many sources.
- Onboard pop filter works very well.
- No need for additional shockmounting.

CONS
- Won't suit every voice or instrument.

SUMMARY
The New Twenties is a striking-looking capacitor microphone that offers stylised looks and a stylised sound. Capable of delivering vibe-heavy sounds on louder sources, or the right vocal, it can also provide more natural results if needed for voice work or acoustic instruments that suit its rolled-off, mid-focused sound.

NEIL ROGERS

Madrid-based Tierra Audio have always done things differently, and since their first release back in 2018 they have built up a range of original, handmade studio outboard products including preamps, compressors and summing mixers. Tierra also take their environmental impact seriously, and their website lists a number of eco-friendly decisions, such as the use of sustainable bamboo for the panels on their outboard units and an emphasis on sourcing parts locally wherever possible.

In April 2022 (www.soundonsound.com/reviews/tierra-audio-flavours) I reviewed their range of inline mic preamps, which have culinary names such as Chilli and Mint and offer 'flavoursome' ways of boosting microphone signals. In this review, we have the Spanish company's first capacitor microphone (they've previously launched a range of active and passive ribbon models), and in keeping with their previous releases, the New Twenties is not your average mic.

The Roaring '20s

Tierra have chosen a retro visual theme for their new capacitor mic, which aims to offer distinctive looks (important in the 'content' age that we live in) along with some innovative technology to achieve a high-quality but characterful sound. The mic is not intended to directly reference the sound of microphone technology available 100 years ago; rather, as Tierra explain, when developing the mic during the Covid lockdowns, they felt inspired by "the tireless spirit that the women and men of the 1920s showed when coming out of the aftermath of World War 1". With a stylistic nod to the elegance and glamour of an era known as the Roaring '20s, this mic is very conscious of its looks, and can even be fitted with custom-engraved 'ridges', for you to add your studio, social media channel or podcast name and logo to. This is not a cheap or gimmicky microphone, however, so I was very intrigued to see if it sounded as good as it looks.

Features

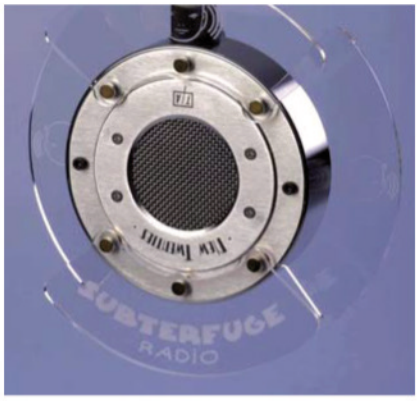
The New Twenties is a cardioid-only, large-diaphragm, electret-capsule microphone, and without providing any further details, Tierra explain that they sourced a USA-built capsule that uses only "the highest-quality materials". The body is primarily a solid piece of stainless steel, which, as well giving a substantial look and feel, protects the internal electronics from electrical interference. High-quality, low-noise FETs are used to raise the signal level from the capsule before it reaches the Lundahl output transformer — which is apparently a key part of how the mic achieves its distinctive sonic characteristics. Tierra have also developed their own method of internal shock protection for the capsule, which sees them incorporate precision 'micro-springs' — of the type typically

used by watchmakers — to suspend the capsule in the mic's body.

Great Expectations

The New Twenties comes in a neat, custom-made box made from sustainable bamboo, with the mic secured inside using magnets. This approach is also used for the excellent onboard pop filter, which has been nicely incorporated into the mic and can be quickly attached or removed from the small but powerful N52 neodymium magnets that hold it. The mic doesn't require any additional mounting, but the down side of this is that it's slightly tricky to get it into any angled position without a mic stand that doesn't have a second boom stage. My first use for the New Twenties was, appropriately enough, capturing a singer-songwriter playing an early 20th Century-inspired song that I thought would be a great first test of this unusual-looking mic. I was keen to capture the artist singing and playing her acoustic guitar with one mic, and after finding a nice spot where it captured a good balance of voice and guitar, it produced a natural reproduction of how the artist sounded in the room, and required very little additional mix processing.

It's hard not to approach a new microphone without some preconceptions, and having seen how Tierra describe the New Twenties on their website, I was expecting something quite different from what I was hearing from the speakers. One of the things I noticed straight away was how lean this mic is in the bottom end: the low end roll-off begins gradually around 100Hz before falling away steeply below 50Hz. This is not a criticism at all and suits many situations perfectly, but I didn't find a single application in my studio in which I would describe this as a full-bodied, 'warm' sounding mic! A great example was when I used it to record my studio's in-house acoustic guitar alongside my more conventional choice of an AKG C414. It was striking how different the low-end responses were, and while this is often an area that you would carve out in a mix, it didn't suggest to me that this mic would suit every job in the studio.



The mic ships with 'ridges', which Tierra can custom-engrave with your studio's logo as a cost option.

The Moment It Clicked

I'm going to admit that after my first few goes with this mic — especially bearing in mind the price — I thought it was turning out to be a textbook case of style over substance. That was before I tried the New Twenties as a vocal mic on two recording session, in which it absolutely excelled. The first was recording a male vocalist singing a soulful falsetto-style vocal, and it was here that I first began to hear coloration clearly being imparted by the mic's transformer. With no outboard in the recording chain other than a neutral-sounding preamp, the vocal immediately had an authentic, vintage-style vibe, with a rolled-off low end and a midrange presence that was just on the edge of becoming saturated.

The next session I tried it on was a jazz/soul-style band playing live together in the room. The band wanted to cut the vocals in the same room as the drums, and the key to this working was for the mic to sound good both on the vocal and, because of the inevitable spill, as a drum room/ambience mic. The New Twenties worked superbly, providing a vibe-heavy vocal sound that suited the style, combined with a flattering drum room sound that was complimentary to the mix. At least two members of the band asked me "What's that mic?" after listening to the playback, and as the session was also being filmed, the combination of looks and sound was perfect.

I also got great results using the mic on an electric guitar cab; in fact, on any louder source that suited a mid-forward, saturated sound it seemed to work really well. As well as attenuating low frequencies, the New Twenties also rolls off very high frequencies, in a similar way to a modern ribbon mic, and this seems to be another subtle factor in how this mic works so well on certain sources.

Another observation worth mentioning, and which I noticed especially when talking into the mic, is how the low-frequency roll-off almost completely negates any proximity effect you would typically get from using a capacitor mic up close.

Hear For Yourself

I've prepared a number of audio examples of the New Twenties, so you can hear how it performed on a range of sources. You can stream and download the files at https://sosm.ag/tierra-new-twenties.

This means it can work well on spoken word, but I would be wary about using it in typical voiceover sessions where clients tend to look for a more full-range sound. For a content provider who wants a cool-looking mic with a bit of a 'sound,' this could be a great option, however.

Summing Up

The strength of the New Twenties seems to be in bringing forward a little of the midrange excitement of a source whilst adding subtle saturation. This, combined with its low-end roll-off, can produce a vibe that is very pleasing indeed. The same qualities can make it a little fussy, however, and I would often have to tuck in a certain frequency with EQ to get it to really work — though this nearly always felt straightforward.

If there's a drawback, it's that the New Twenties arguably sits a little above its natural price point, given that it might not work equally well on everything. I think serious content creators and studios looking for a vibe mic are happy to pay a decent amount for a product that is well-made and looks the part, but in the $1000 category, you have the choice of a lot of well-known mics with additional features. That said, part of the reason for the price is Tierra Audio's commitment to their environmental ethics: like many things in our society the right choices are not always the cheapest, and you can only applaud a company trying to make these kinds of decisions work.

In any case, this is a great-looking, handmade mic that, on the right voice or instrument, absolutely delivers the goods sound-wise. Once I understood what its vibe was I found this to be a super addition to my more conventional mics, and with the caveat that you should try it out first, this could be a great option for a visual content-based setup.

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MODULAR

Cosmotronic Cosmix Pro

Eurorack Module

We live in a world of component shortages and economic uncertainty, where boutique developers are often found on the front line in the fight for survival and must therefore make every design count. Dutch developer Cosmotronic — aka the one-man operation of Matthijs Munnik — has relatively few modules on the market yet maintains a laudable strike rate, with a focused range of designs bridging the gap between reliable and innovative with aplomb. The Cosmix Aural Enhancing Stereo Mixer was one module that certainly fit this description, now discontinued due to a lack of components for its much-loved dual-band distortion circuit. This led Matthijs to a fork in the road: double down on the Cosmix and find alternative components to try and recreate it, or make lemons into lemonade and go in another direction entirely? Thankfully for us, he opted for the latter: the Cosmix Pro is an eight-channel mixer offering a broad range of routing possibilities, with generous amounts of visual clarity and several functions usually reserved for much larger mixers.

In a sense, the Cosmix Pro toes the line perfectly between an upgrade and a lateral shift; by this I mean it can be paired with the Cosmix via a rear Link cable, so even if you're an owner of its predecessor there's something for you here. It has 6HP on the width of the Cosmix, and while it offers the same amount of inputs, not only does it do more with them, its spacious layout feels clearer, faster and, well, more like a 'real' mixer — hard as that notion is to qualify.

The Cosmix Pro's panel presents a total of six channels, constituting four mono and two stereo channels. Each of these has a dedicated slider and miniature three-colour VU meter, as well as a large mute button and a pan pot. A master level knob sits on the bottom right of the module beneath a pair of master meters. So far so solid. The mute buttons are brilliantly designed: clickless and brightly lit, they are as well suited for bringing parts in and out of a performance in a low-light setting as you'll find in a Eurorack module. They also engage on release and not on contact, meaning it's easier to bring things in and out right on

cue. Beyond this is where things get a little more interesting — or, should I say, 'pro'. The first four channels each offer a low-cut switch, which applies a tasteful 1-pole high-pass filter at 80Hz. Very useful for staving off low-frequency build up in a patch, particularly if you have a few low-pass filters on the go. The two stereo channels, meanwhile, each have a gain switch to boost their respective incoming signals by 18dB; in other words, to bring anything line-level up to modular level. This immediately opens up the Pro to the world outside of your system, unifying it with anything from samplers to laptops to centralise your setup, on stage or in the studio.

Where the original Cosmix had one mono aux send, not only does the Pro boast two sends, but one of them is in stereo. Now luscious stereo reverbs or ping-ponging delays can be applied to any input signal, mono or stereo, leaving the mono aux free for things like feeding a side-chain input on a compressor, to name but one possibility. The mono aux 1 is reserved for the first four channels, while aux 2 is available for all six. Capping it all off are two CV inputs for the panning on stereo channels five and six, whose pan pots become stereo balance controls when both sides are connected, opening up some interesting opportunities for movement across the stereo image.

There are one or two things I'd have loved to see on the Cosmix Pro, for instance pre-fader aux sends, or at least the option to set them so. I'm the kind of person who keeps my aux sends permanently set to pre-fader, which not only gives me more flexibility with the wet/dry balance on my effects sends but also allows me to use the auxes as alternative outputs for certain channels if needed. While the Cosmix Pro's sends are unalterably set at post-fader level, I should say that with its generous amounts of headroom across the board, with my main channel fader low and aux level turned up high, I wasn't far off being able to hear my effects fully wet. I would also have appreciated a headphone

output, which doesn't feel like a stretch considering the Cosmix Pro definitely has the panel space. I've seen the question of quarter-inch outputs asked elsewhere, but I believe mini-jack is the right call for any Eurorack mixer; not least because it lets you easily feed the main output into other modules. At the end of the day, to adapt GK Chesterton's well known adage: mixers, like art, consist of drawing the line somewhere. One could go on forever thinking up potential features — inserts, EQ, polarity buttons and more — but this would be inappropriate. Ultimately, a Eurorack mixer has to be selective with its functions, otherwise you may as well just get a standalone mixer. The Cosmix Pro has chosen well. It's sleek, it's smart, it's reliable, it's creative (it's also very cute), and suffice to say I'm struggling to imagine my system without it. *William Stokes*

$ $399

W www.cosmotronic.nl

Xaoc Devices Sofia

Eurorack Module

You've heard 'never judge a book by its cover', but how about 'never judge the gear by its faceplate'? It's a principle I must constantly remind myself of in the studio; by this I mean that so many guitar pedals, plug-ins and, of course, Eurorack modules look and feel so lovely (and occasionally so hilarious) that it's actually easy to forget the most important thing: the sound. In fact, it's now got to the point where the worse or more boring something looks, the more inclined I am to see what it can do — and inversely the prettier it is the more sceptical I am.

Needless to say, then, My eyebrows were raised upon unboxing the Sofia from Polish developer Xaoc (pronounced Chaos) Devices. Even the module's full name is suspiciously pretty: the Sofia Transcendent Waveform Analog Oscillator Model Of 1955. Putting the lovely included Xaoc Devices red tote bag and paper manual to one side, I picked up the module and tested its knobs and switches, as I'm in the habit of doing. A perfectly weighted stepped dial for octaves above a large, classic-looking frequency knob, four sliders, six smaller knobs, all beautifully firm and smooth to the turn. A gorgeously designed silver and black panel with a vintage look that would fit right in at the venerable PRES (Polish Radio Experimental Studio)... It's all just a little too lovely, right? Wrong. From the moment I mounted it and powered up my system it was clear that this module sounds every bit as good as it looks and feels, and then some.

So, what actually is the Sofia? And what makes it transcendent? Well, it's an oscillator. And almost certainly unlike any other oscillator you've tried. At the heart of the Sofia is something called *fonction d'onde formantique*, or FOF. FOF synthesis is essentially the synthesis of vocal-like formants without filtering, achieved by superimposing decaying sinusoidal tones onto each individual cycle of a fundamental base waveform. In this way it ostensibly mirrors what the Sofia manual describes as the "parallel acoustic resonant filters that produce decaying sinusoidal tones in response to each pulse of air pressure from the larynx". In other words, the way a human voice behaves to achieve complex waveforms. Viewed on an oscilloscope, these decaying sinusoidal tones (I can't help but feel that term sounds like some kind of nasal infection) look like waves 'rippling' through each cycle of the base wave, which is why they have come to be known, at least for the purposes of the Sofia, as 'ripple

elements'. The Sofia generates two such ripple elements that both work on the same sine base wave. Their characteristics can be adjusted, multiplied and modulated in all manner of ways, and their harmonic contributions can be mixed in any combination — that is, relative to each other and relative to the base wave, which is itself slightly saturated for that extra bit of character.

While this workflow takes some getting used to, it's a wonderful way to go about sculpting sounds, and at the end of the day it sounds, frankly, excellent. I love Xaoc's description of the module's palette in the manual: "a wide range of acoustic, woody, organic and animal-like, as well as high, fuzzy and bright sounds". Couldn't have put it better myself. Even when patched to simply drone away, moving various parameters from one extreme to the other creates three dimensional movement that at times sounds like a toothy pair of sync'ed oscillators, at others like 303-style resonant filtering, and at others still like some kind of bubbling, frozen vocoder.

It achieves this range with surprising simplicity. Beyond the aforementioned octave dial and frequency knob, the Sofia's panel offers three controls for each of the two ripple elements, all with accompanying CV inputs. There are two controls in the first instance: one to blend between the fundamental tone and the ripple elements with all their wondrous overtones; and one to control the balance between ripple elements A and B. Beyond this, both ripple

» elements have identical controls: Ratio, which adds cycles (and therefore harmonics) to the ripple wave; Damp, which controls the decay rate of the ripples' amplitude through each cycle of the fundamental tone; and Warp, which alters the density of the ripple wave: the ripples can get denser as they move through time, or they can get sparser.

This, of course, is just the theoretical side of things. There's no way you can track these effects in detail just by listening to them — at least, not in the way you can with a conventional envelope, since everything is repeating at audio rate. It's therefore well worth keeping an oscilloscope to hand if you have one, at least in the beginning, just to keep tabs on what's actually going on. It's also fascinating, to be honest. The main thing, of course, as we have previously ascertained, is how it sounds. So it hence became a simple case of exploring with my ears how each of these controls has its own way of squeezing, stretching and twisting the sound of Sofia. On top of this, it's possible to switch between sine and square waves for the ripple elements, square of course being more angular and harmonically rich. Things really come alive with modulation, and self-patching is encouraged. Even the simplest of LFOs coaxed out tones ranging from visceral to ethereal; add some reverb, and wow!

The cherry on top of such harmonic richness is the fact that the Sofia offers a whopping six possible outputs. Unite any combination of these with a mixer and you can achieve astonishing width and movement, for example keeping the fundamental wave down the middle of the stereo image and panning the ripple elements left and right. Fantastic. An astonishingly powerful and flexible module, the Sofia has absolutely become one of my favourite oscillators and I would recommend it to anybody. Whoever said looks could be deceiving?

William Stokes

$ $490

W www.xaocdevices.com

Sub-$200 Function Generators

Modular Round-up

The humble function generator: with the simple task of producing LFOs and envelopes — at least, at its most basic level — it is an oft-overlooked but key part of any system. While generally considered part of the (less fun, if we're honest) 'utility' family of modules, alongside things like VCAs or mults, many developers are, you might say, pushing the envelope (sorry) with their function generator designs to produce some very interesting circuits. Of course, modular is an expensive craft: one might save with anticipation to buy an exciting sampler or drum module, but it's generally with reluctance that we invest in utility modules, despite the fact that they're essential for those other modules in our system to work at their best. Not all function generators need to break the bank, though. Here are some of our favourite function generators that will capably augment your patches, all weighing in at well under £200.

ALM/Busy Circuits Pip Slope (Rev II)

British developer ALM/Busy Circuits' Pip Slope fits an impressive amount of enveloping goodness into a miniature 4HP. Building on the original Pip Slope, its CV-controllable attack and decay times boast a whopping range, from 1ms to 7 minutes, and it also has the ability to morph smoothly between exponential, linear and logarithmic slopes via a Shape knob. Perhaps most interestingly, it also features a Loop mode that can cycle the envelope with decreasing amplitude or decreasing time if desired, opening the door for delay-type effects or sweeping dynamic movement across a patch. At just over the $100 mark, this is certainly bang for your buck.
www.busycircuits.com

Endorphin.es Airstreamer 4

We love Endorphin.es' aesthetic — somehow they even manage to make a function generator look cool. Condensing the previously 6HP Airstreamer into a Pip Slope-rivalling 4HP, the Airstreamer 4 selectively offers CV control over its decay time, though not its attack time. This might sound limiting, but there's masses here: dual envelope CV outputs offer unipolar or bipolar voltage ranges, while a polariser knob can attenuate or invert its output voltage, also providing an additional VCA input to scale and invert the envelope for a "compact side-chain envelope solution", in Endorphin.es' words. There's even a very smart Volt-per-octave input for tuned control when oscillating at audio rates. www.endorphin.es

After Later Audio Tilt

After Later Audio's take on a five-decade old Serge classic, the Slope Generator, the Tilt is described by its developer as "really a jack of all trades." Like its predecessor, it offers both unipolar and bipolar outputs, adding separate controls for exponential and logarithmic responses at both the rise and fall stage, as well as a gate/sustain function for a full ADSR envelope. Usefully, there are both inputs and outputs for both the rise and fall stages of the envelope, as well as one for both. All the key functions you'd hope for are here, including things like the capacity to use the In input to create a slew limiter, and in a rather lovely black and gold finish, too.
www.afterlateraudio.com

Erica Synths mki x es.EDU EG

The cheapest on our list at just $55, this envelope generator was brought forth by the excellent mki x es.EDU, a collaboration between Erica Synths and educational electronics whizz Moritz Klein, and it seeks to coax even the most timid prospective DIY-er into the world of electronics and circuit design. Described as a "slightly optimised" ADSR envelope generator, the EG is a simple and useful function generator, configurable to either an ASR or an AD response that ignores input gate time. A toggle switch for looping and an inverted output cap off an interface that's all about the essentials. A simple kit at a great price, resulting in a great module. What's not to love? Best order that soldering iron. www.ericasynths.lv

Make Noise FUNCTION

Derived from one of Eurorack's most successful function generators — make that one of the most successful modules — the Make Noise Maths, the Function's staggering time range begins at 2ms and extends to a possible 25 minutes per cycle, for those truly slow-growing sound experiments. Beyond this is a flexible palette of linear, logarithmic and exponential functions, as well as inversion and integration for lag, slew and the like. A Hang input allows the circuit, as Make Noise put it, to "stop dead in its tracks" for capable sample and hold, track and hold, stepped movement and more.
www.makenoisemusic.com

William Stokes

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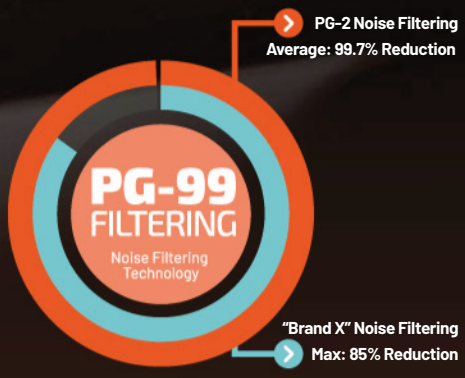

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Apple MacBook Pro & Mac mini

Personal Computers: Part 1

We begin an in-depth, two-part examination of Apple's M2 Pro and Max chips, and the new MacBook Pro and Mac mini computers that house them.

MARK WHERRY

It's been nearly three years since Apple announced the transition of the Mac product line to an architecture based on the company's own custom silicon. And, of the three such brain transplants in the Mac's history — the first being the shift from 68k to PowerPC in 1994, followed by the move to Intel's x86 architecture beginning in 2005 — the transition to Apple Silicon has arguably been the most prosperous.

Having created the A-series chips used in iPhones and iPads, where an entire system is engineered into a so-called system-on-a-chip (SoC), Apple extended the approach to the Mac with the M1, unveiled in December 2020 alongside three new Macs retrofitted with this new hardware: the Mac mini, the MacBook Air, and the 13-inch MacBook Pro. It's not an understatement to say that the M1 didn't just meet expectations, it shattered them.

Compatibility with existing applications and plug-ins was largely seamless, and even endlessly needy audio applications coded for Intel chips were able to meet professional requirements when running under Rosetta 2's binary translation. But with newly native upgrades to products such as Apple's own Logic Pro leading the charge, for creative professionals it was clear that Macs based on Apple Silicon really were going to be a step forward, representing more than merely the sum of the of the many technologies being deployed.

Moreover, performance was only one side of the equation. The M1 was able to do its work with remarkable efficiency, extending battery life and largely eliminating fan noise from the list of complaints troubling the audio user.

Memory Holes

The M1 did, however, have one unavoidable down side for musicians and audio engineers working with large projects and sample libraries: memory. Unlike previous Macs, the M1 employed what Apple refer to as Unified Memory, which was shared between the CPU and GPU cores (and other domain-specific engines) without redundancy or overheads. However, to fully take advantage of this approach with the minimum amount of latency, the memory was included in the same package as the SoC itself, and the M1 was only available in two configurations, with either 8GB or 16GB of LPDDR4X unified memory, which couldn't be upgraded after purchase (not that this is anything new for Mac users!).

And so, in October 2021 — less than a year after introducing the M1 — Apple launched two new chips extending the capabilities of that first offering: the M1 Pro and the M1 Max. Four different 'core' configurations were disclosed, with the M1 Pro supporting a maximum of 32GB LPDDR5 unified memory with over twice the bandwidth at 200GB/s, and the M1 Max doubling these limits again with 64GB memory and a bandwidth of 400GB/s. However, rather than deploying these new chips in desktop computers, Apple launched them with brand-new

Among the first machines to use the new M2 Pro and Max chips were Apple's latest MacBook Pro laptops.

14- and 16-inch MacBook Pro models, which delivered an unprecedented blend of performance and efficiency.

Apple unveiled the final member of the M1 family, the M1 Ultra, in March 2022, with the introduction of the desktop Mac Studio. Available in two base configurations, featuring either an M1 Max or the new M1 Ultra, the Mac Studio has in many ways proved to be this generation's iMac Pro: a product to keep demanding creative professionals happy while Apple figure out how to proceed with a new Mac Pro.

The M1 Ultra is particularly interesting, as it comprises two M1 Max chip dies in a single package, which communicate with each other via a new interconnect technology Apple call Ultra Fusion. This previously unannounced feature uses a silicon interposer to act as a kind of glue that makes the M1 Ultra appear to the system as single chip. This means the M1 Ultra has, quite literally, twice the specifications of the M1 Max, supporting a system with a maximum of 128GB unified memory and 800GB/s memory bandwidth — although, in terms of capacity, this still falls short of the 768GB or 1.5TB limits of the still current, Intel-based Mac Pro models.

Moving On Up

One didn't need to be a soothsayer to predict that, following the M1, Apple's next-generation Mac chip would be named M2 and adopt the same Pro and Max variations. Where the M1 represented a revolution, the M2 is more of an evolution, refining and polishing every jot and tittle of the system. The 'Firestorm' performance and 'Icestorm' efficiency CPU cores (first seen in the iPhone 12's A14 Bionic chip) were upgraded to the newer 'Avalanche' and 'Blizzard' performance and efficiency cores (which debuted in the iPhone 13's A15 Bionic chip). These newer CPU cores are clocked slightly higher than their predecessors, with a base frequency of 3.42GHz vs 3.2GHz in the M1, and offer several improvements, such as a larger Level 2 cache for the performance cluster. As with the original M1, the M2 is an eight-core chip featuring four performance and four efficiency cores.

Turning as always to Primate Labs' trusty Geekbench (see box), the M2 chip in the current 13-inch MacBook Pro scores 2606 and 9860 in the single and multi-core tests respectively, which showed improvements of 10 and 17 percent over the M1 chip in the previous 13-inch MacBook Pro, which scored 2356 and 8426 in the same tests.

The M2 addressed many of the M1's shortcomings, particularly by adopting LPDDR5 memory, raising the memory ceiling to 24GB, and increasing the memory bandwidth by 50 percent from 66.67GB/to 100MHz. Even so, demanding users — like, say, musicians and audio engineers — were eagerly awaiting the expected, more powerful iterations. Apple duly began this year by announcing the M2 Pro and M2 Max, accompanied by revisions to the Mac mini and the 14- and 16-inch MacBook Pro models.

If you followed the development of the M1 to the M1 Pro and M1 Max, Apple have used a similar playbook in creating the M2 Pro and M2 Max. For example, as with the M1 Pro, the M2 Pro is available in two configurations with either six or eight performance cores, while the M2 Max, like the M1 Max before it, has eight performance cores. However, unlike the M1 Pro and Max, which both offered two efficiency cores, the M2 Pro and Max feature four such cores, meaning that, where the M1 Pro and Max were either eight- or 10-core chips, the M2 Pro and Max are 10- or 12-core chips.

If we look at the single-core results first, the M2, eight-core M2 Pro, 10-core M2 Pro, and M2 Max chips score »

Apple 14-inch & 16-inch MacBook Pro (2023)

From $1249

PROS

- Everything that was great about the design of the latest 14- and 16-inch MacBook Pros remains the same. The gorgeous display, the speaker system, the battery life, and so on.
- For those who find more never enough, especially when in a portable form factor, the M2 Pro and M2 Max bring refinement and polish in all domains, evolving the overall performance.
- More than 64GB RAM is available in a portable Mac — finally!

CONS

- Can quickly become expensive when configured with the higher-end chip, memory and storage options, which are set in stone at the point of purchase.
- The lower-capacity SSD configurations offer less read and write throughput than the previous M1 Pro/Max-based models.

SUMMARY

The 14- and 16-inch MacBook Pros were already fabulous machines, giving musicians and audio engineers true desktop-class performance in a precisely engineered notebook. For those who need the absolute pinnacle of portable performance, these new M2 Pro and M2 Max-based models will elevate you to that level if your budget permits.

Thunderbolt 4

Thunderbolt 4 offers the same 40Gb/s bandwidth as Thunderbolt 3, but doubles the minimum bandwidth for PCIe data from 16Gb/s to 32Gb/s. This means that those using NVMe-based storage devices (since NVMe utilises PCIe for transport) should see greater performance, as will anyone who relies on an expansion chassis for PCIe-based audio and video cards, not to mention higher-end networking and storage cards. And the 40Gb/s transfer rates are guaranteed with the use of Universal Thunderbolt 4 cables up to 2m in length, which were available for less than $60 via Amazon at the time of writing.

Other improvements courtesy of Thunderbolt 4 include the mandate that a minimum of two external 4K displays must be supported, and that external devices connected via a dock must be able to wake your computer. And speaking of docks, with Thunderbolt 4 supporting accessories with four Thunderbolt ports, you should start to see docks with additional Thunderbolt ports, which is where that extra bandwidth for PCIe data should come in rather handy.

» much the same: 2606, 2556, 2667 and 2756 respectively, which is to be expected since the same 'Avalanche' and 'Blizzard' CPU cores are at the heart of each chip. However, if we instead consider the multi-core results, the performance differences between the chips become more apparent. The M2 (with four performance cores), the M2 Pro (with six and eight performance cores) and the M2 Max (also with eight performance cores) score 9860, 11830, 14349 and 14390 respectively.

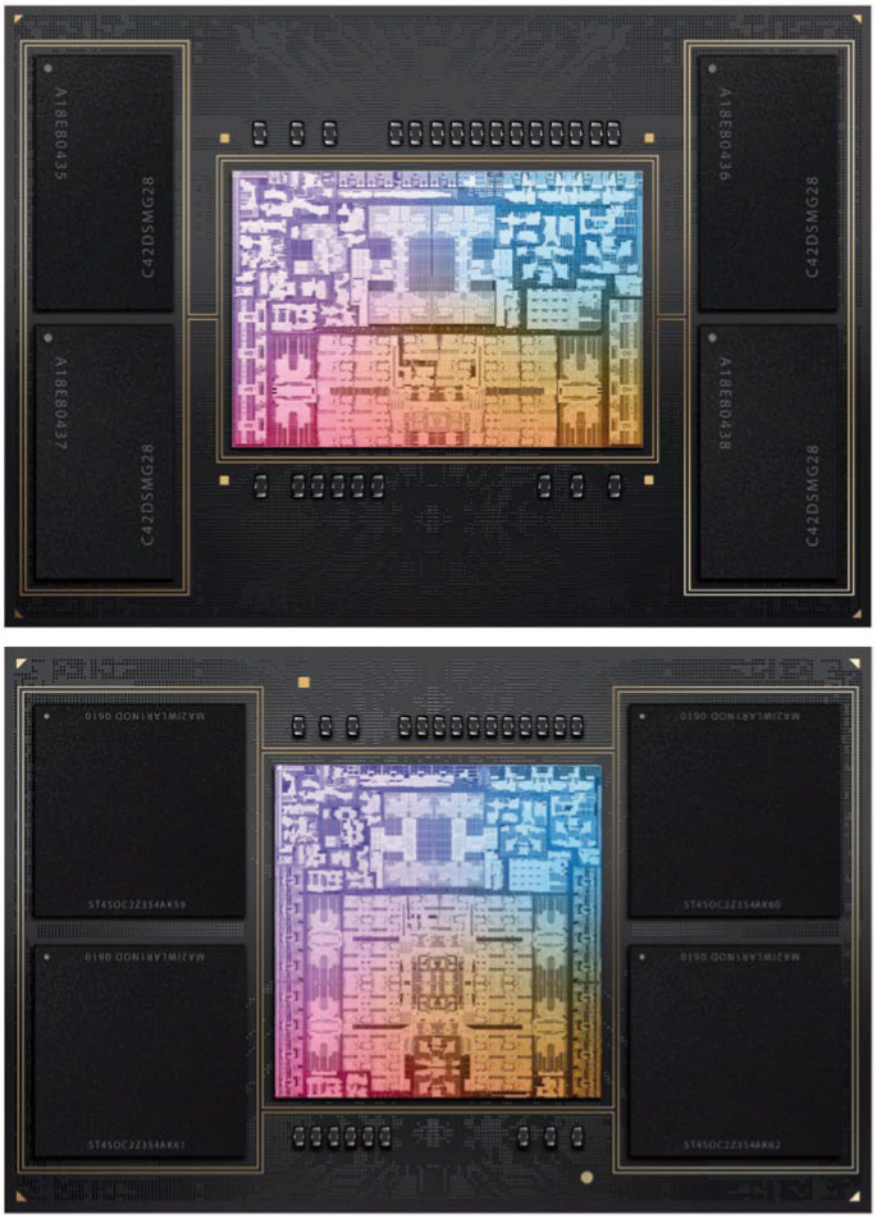
As you would expect, the CPU performance between the 12-core M2 Pro and the M2 Max (which both offer the same configuration of four efficiency and eight performance cores) is equipollent. It's worth mentioning, though, that the M2 Max in the 16-inch MacBook Pro allows a high-performance mode to be engaged in the Energy Saver System Settings, which gives a boost to the base frequency of the chip. This setting isn't available when the M2 Max is configured with the 14-inch MacBook Pro. Interestingly, although the M2 Max boasts double the memory bandwidth of the M2 Pro, this doesn't make much difference to the outcome of Geekbench's tests.

It Had To Be GPU

Even if you're going to stick with a 32GB configuration, GPU performance might be a reason to consider the M2 Max over the M2 Pro (see box). Video and graphics applications naturally benefit from hardware acceleration, but the GPU isn't wasted on music and audio software, since developers are embracing Apple's Metal API for accelerated graphics in rendering user interfaces, and harnessing its programmable shaders for audio processing. As an example, Steinberg have done a tremendous job in using Metal to reduce the CPU overhead of drawing an application's user interface, as we shall see, while GPU Audio plan to do just what their name suggests.

Each member of the M2 family provides higher GPU core counts than its direct predecessor. The M2 has a 10-core GPU to the M1's seven or eight cores; the M2 Pro is available with either 16 or 19 cores, compared with 14 or 16 in the M1 Pro; and the two M2 Max options feature either 30 or 38 cores, as opposed to 24 or 32 in the M1 Max. Testing the new M2 and M2 Pro chips with Geekbench 6's Compute test (with the M2 Max results gleaned from the Geekbench browser), the 19-core M2 Pro GPU yielded a 17 percent increase over the 16-core M1 Pro, and the M2 Max's 38-core GPU showed a similar increase over the M1 Max's 32-core equivalent.

At present, though, GPU performance alone probably doesn't justify upgrading »



The M2 Pro and M2 Max chips both have four efficiency cores, but the Max (lower) has eight performance cores to the Pro's six.

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Apple Mac mini (2023)
From $599

PROS
- The base model is possibly the most affordable Mac ever.
- Thunderbolt 4 streamlines connectivity, enabling multiple displays to be officially supported.
- The maximum amount of memory has been doubled to 32GB in the M2 Pro model, but...

CONS
- ...this limit is still half the 64GB maximum of later Intel-based models released in 2018.
- High-end configurations can quickly become as expensive as the cheapest, M1 Max-based Mac Studio.
- Some front-panel connectivity, as with the Mac Studio, would be the cherry on the icing on the cake.

SUMMARY
With two system-on-chip configurations available, the new Mac mini is both more affordable and considerably more powerful, with both models benefitting from the media engine's performance for video encoding and decoding. There's really very little to complain about.

The Mac mini has been a stalwart of Apple's line for many years, and provides the most affordable point of entry into the Mac ecosphere.

The M2 Max variants of the Mac mini (above) benefit from double the number of Thunderbolt 4 ports.

» to a higher-spec model if music and audio are your main pursuits. The GPU resources in any of the M-series chips are more than adequate for rendering user interfaces, and even GPU Audio's offerings are still unproven at scale and are currently in the public beta stage.

More Mac Than Mini

The Mac mini is in many ways the dark horse of Apple's Macintosh range. Since its introduction at Macworld in 2005 as an affordable, entry-level 'BYODKM' (bring your own display, keyboard and mouse) system, the Mac mini became ever more capable during the Intel years, enabling and reaching a broader number of applications and audiences. Between 2009 and 2014, Apple even offered the Mac mini in a server configuration for those not requiring enterprise-level performance.

Because Intel-based Macs could run Windows natively thanks to Boot Camp, they've been put to all sorts of uses in studios. Anything from running software for touchscreens, other control surface or MIDI processing-related tasks to file-sharing, portable Vienna Ensemble Pro servers, or even a basic workstation can be done with a Mac mini running Mac OS or Windows or both. And it's perhaps because of this that Apple maintained an Intel-based Mac mini in the product line alongside the M1 model — until now.

The new M2- and M2 Pro-based Mac minis replace both previous models and look identical, retaining the same silver veneer as the Mac Studio. In fact, the Mac mini can almost be thought of as a junior Mac Studio. There's likewise no visual clue to any changes in connectivity. You'll find the same power button and IEC C7 connector, and I feel that Apple's engineers deserve some credit here for keeping the Mac mini's power supply internal. Computers with similar form factors, such as HP's Z2 Mini G5 or G9 »

Pro Vs Max

Given the CPU equipollence between the 12-core M2 Pro and the M2 Pro Max, as detailed in the main text, you might be wondering why it could be necessary to budget more for the latter than the former. And there are essentially three reasons: memory (including memory bandwidth), the capabilities of the media engine for video work, and GPU performance, which is described in the main text.

Memory (and memory bandwidth) is perhaps most important of these three factors for musicians and audio engineers. While the M2 Pro can be configured with either 16 or 32 GB (with a memory bandwidth of 200GB/s), the M2 Max doubles the memory bandwidth to 400GB/s, with either 32GB or 64GB in the base model. And although these numbers are the same as the previous M1-based generation, a new high-end M2 Max configuration is available with 96GB unified memory — if you can afford the additional $1500 cost on top of the base model.

Another reason to consider the M2 Max over the M2 Pro, which is probably less important to musicians and audio engineers, is the media engine. Formalised in the M1 Pro, the media engine provided hardware-accelerated encoding and decoding for H.264, HVEC, ProRes and ProRes RAW video, comprising a video encode engine, a video decode engine, and a ProRes encode/decode engine. The M1 Max extended these processing capabilities by offering two video encode and ProRes encode/decode engines.

The M2's media engine has been enhanced with support for 8K video, and all three M2-series chips now feature an implementation of this component, with the M2 Max once again offering additional engines. For video playback, most users will probably find the M2 Pro's ability to play back up to 23 4K or five 8K streams of ProRes-encoded video sufficient. However, if your video-editing needs require additional heft, the M2 Max's second ProRes accelerator makes it possible to play back up to a staggering 43 4K or 10 8K ProRes video streams.

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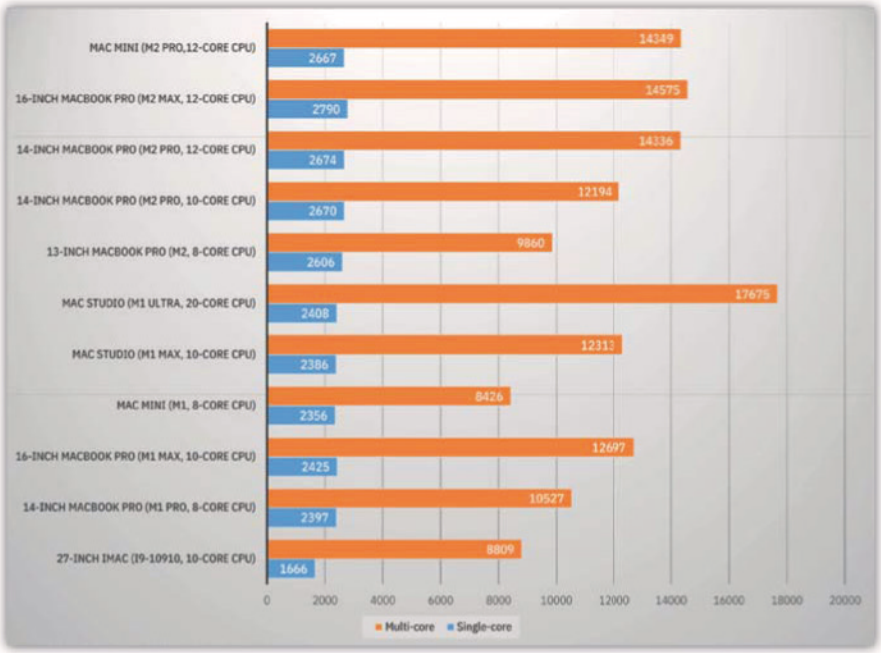

The single- and multi-core results for Apple Silicon-based Macs. Note that the last Intel-based 27-inch iMac only narrowly beats the multi-core score for the M1-based Mac mini, falling comfortably into last place in terms of single-core performance.

Here are the Metal scores for various systems tested with Geekbench 6's Compute test. While the Mac Studio, with its higher GPU core count, is still the best-performing system, you can see the bolstered GPU hardware used by the M2 family of chips easily outperform the previous generation. Note that the M2 Pro-based systems on review easily surpass AMD's very capable AMD Radeon Pro 5700XT graphics hardware used in the last Intel-based 27-inch iMac from 2020.

» models, all require external power bricks, as does Microsoft's own ARM-based Windows Developer Kit (previously known as Project Volterra).

Alongside the power connector, there's Gigabit Ethernet (which is configurable to 10 Gigabit Ethernet, as before, for an extra $100), Thunderbolt and HDMI connectors, not forgetting two USB-A ports and a 3.5mm headphone jack. As on the Mac Studio, the power button and the headphone jack are on the rear of the machine, which is inconvenient, but perhaps necessitated by the ergonomics of the design.

Thunder & Lightning

However, just because the ports at the rear of the new Mac mini look the same as before doesn't mean they are functionally the same. For example, where the preceding Mac mini offered Thunderbolt 3 with support for USB 4, the new model brings Thunderbolt 4 support (see box), as found on other recent Apple Silicon Macs. And, best of all, if you opt for a new Mac mini with an M2 Pro chip, this system is equipped with four such ports rather than two, as on the M2 or previous M1-based models.

In addition to providing Thunderbolt 4, these USB-C-style ports also support DisplayPort, USB 4 (up to 40Gb/s) and USB 3.1 Gen 2 (up to 10Gb/s), as well as Thunderbolt 2, HDMI and VGA via adaptors available from Apple or other third-party vendors. I had no problem using the Mac mini's 10Gb Ethernet to connect with a Pro Tools | Carbon interface, even when running a Session at 192kHz, whilst linking to a network via a USB-C hub that offered an additional, independent Gigabit Ethernet connection.

Like every Mac Apple have introduced since October 2021, the Mac mini's headphone jack features what Apple refer to as "advanced support" for high-impedance headphones. This has been a welcome change for those wanting a better out-of-the-box headphone experience without having to connect a separate audio interface, and it sounded fine with a trusted pair of Audeze headphones.

One crucial difference compared with the previous M1 Mac minis concerns support for external displays. The »

Bench Press

Just as I was finishing this review, Primate Labs kindly released a major new version of the widely used, cross-platform system performance benchmarking tool Geekbench. Geekbench 6 supersedes the previous version 5, which was introduced in September 2019, and introduces both new and updated real-world tests, enhancing its arsenal of workloads and data sets. One of the most significant changes is an overhauled method for benchmarking multi-core performance, measuring how cores cooperate to complete a shared task rather than simply assigning individual tasks to different cores as before. This means that the results offer a better reflection of real-world performance on heterogeneous systems like Apple Silicon, which feature different core types for performance and efficiency.

Therefore, all the test results referenced in this article reflect the newer scoring system as set by Geekbench 6. As do any results quoted from the Geekbench Browser database.

Geekbench 6 can be downloaded from geekbench.com, and a single-user professional licence for Mac OS, Windows, and Linux is priced at $99.

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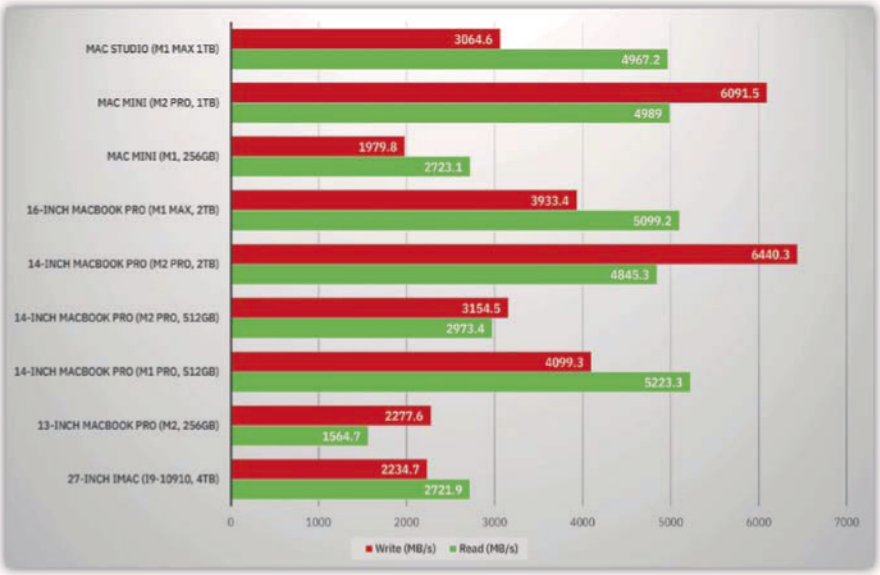

This chart shows read and write speeds for different internal storage capacities on a selection of Macs, using Blackmagic's Disk Speed Test utility.

» M1-based Mac mini officially supported one additional 6K display at 60Hz via Thunderbolt and another 4K display at 60Hz using HDMI, but the new M2 model supports one 6K display via Thunderbolt and either a second 5K display via Thunderbolt or a 4K display over HDMI as before, all at 60Hz. This means you can connect two 5K Studio Displays, or a Pro Display XDR and a Studio Display — although these would be more expensive than the Mac mini driving them!

The M2 Pro-based Mac mini goes further, making it possible to connect two 6K displays at 60Hz via Thunderbolt and one 4K display at 60Hz over HDMI. This would be ideal for media-oriented musicians and audio engineers, who might desire a two-display system for managing editing and mixing windows, plus a dedicated 4K display for video/picture content. Alternatively, this Mac mini also supports one 6K display at 60Hz using Thunderbolt and one 4K display at 144Hz over HDMI, or a single 8K display at 60Hz or a 4K display at 240Hz via HDMI.

The new M2 MacBook Pros have improved options for connecting external screens, with the M2 Max versions supporting up to four 6K displays over Thunderbolt plus an HDMI 4K display.

Support for these higher resolutions and refresh rates would suggest that the M2 Pro model implements HDMI 2.1a with DSC (Data Stream Compression). Apple don't make mention of this specifically, although the M1-based predecessor did quote support for HDMI 2.0 regarding the output of multi-channel audio.

Putting The Pro Into MacBook Pro

Aesthetically speaking, there's nothing to distinguish the new MacBook Pros from the models first introduced just over a year earlier. This is to be expected, given that the 2022 variants benefitted from a complete redesign, and the craftsmanship deployed remains incredible. The display is outstanding, and while the 'notch' caused some minor consternation for some, over a year later I haven't really thought much about it until writing this sentence. And sonically, the response of the in-built speakers continues to impress.

As with the Mac mini, the complement of connectors offered by the 14- and 16-inch MacBook Pros also remains the same. The reintroduced MagSafe connector still provides safely detachable power, although, as I remarked when reviewing the first iteration of these MacBook Pro models, I still feel it's a missed opportunity not to incorporate Ethernet in the power adaptor, as Apple did with the M1-based iMac.

The M2 Pro brings the same improved support for external displays as in the Mac mini. This means you can connect two external 6K displays at 60Hz over Thunderbolt, and another 4K display at 144MHz via HDMI; or, alternatively, you can attach one external 8K display at 60Hz using Thunderbolt or a 4K display at 240Hz over HDMI. Should you have more demanding requirements, opting for a MacBook Pro with an M2 Max affords even more flexibility, making it possible to attach up to four external 6K displays at 60Hz via Thunderbolt, with a further 4K display at 144Hz over HDMI. And again, alternatively, you're able to connect three external 6K displays at 60Hz using Thunderbolt, and one addition 4K display at 240Hz over HDMI.

Test Spec

Apple Mac mini with M2 Pro (12 CPU cores, 19 GPU cores), 32GB unified memory, 1TB SSD, and 10GB/s Ethernet.
Apple 14-inch MacBook Pro with M2 Pro (12 CPU cores, 19 GPU cores), 32GB unified memory, and 2TB SSD.
Tested with Mac OS Ventura (13.2.1) and Apple Studio Display.

Drive Time

The new Mac mini and MacBook Pro models have the same minimum and maximum storage capacities as the previous generation, which are in accordance with the other Macs in Apple's portfolio. M2-based systems, such as the new base Mac mini model, start at 256GB and can be configured with either 512GB, 1TB or 2TB SSDs. M2 Pro-based systems (the Mac mini or MacBook Pro) come with 512GB as standard, with 1TB, 2TB, 4TB or 8TB options available, while the M2 Max-based MacBook Pro is supplied with 1TB and can similarly be ordered with 2TB, 4TB or 8TB SSDs.

Although Apple are to be admired for keeping the starting price low for the base Mac mini model, 256GB really is a paltry quantity of storage for a modern system, especially when Mac OS itself requires just over 13GB of the available space. And with systems like the Mac mini and MacBook Pro we're evaluating in this review, the question of how much internal storage you'll need is complicated by the performance dictated

Previous Apple Silicon Reviews In *SOS*

M1-based Macs (13-inch Macbook, MacBook Air, Mac mini, & iMac): August 2021.
W www.soundonsound.com/reviews/m1-music

M1 Pro & M1 Max-based 14- & 16-Inch MacBook Pros: January 2022.
W www.soundonsound.com/reviews/apple-macbook-pro-m1-pro-max

Mac Studio: May 2022.
W www.soundonsound.com/reviews/apple-mac-studio

There's currently no M2 version of the Mac Studio, but it may still be an attractive alternative to a higher-end Mac mini configuration.

by the SSD configuration. For example, in my own tests using Blackmagic's Disk Speed Test, the original M1-based Mac mini with 256GB of storage had read and write time of around 3000MB/s and 2000MB/s respectively, but in tests reported by Macrumors (www.macrumors.com/2023/01/24/m2-mac-mini-256gb-slower-ssd), a new, M2-based Mac mini with 256GB storage only achieved read and write speeds of around 1500MB/s.

The M2 Pro-based Mac mini model I had available for review, endowed with 1TB storage, achieved read and write rates of 4989MB/s and 6091.5MB/s respectively. And it perhaps goes without saying that this is rather decent. Similarly, the comparably specified 14-inch MacBook Pro with an M2 and 1TB storage reported read and write rates of 4845.3MB/s and 6440.3MB/s. However, while my own, older 14-inch MacBook Pro with an M1 Pro chip and 512GB storage reported read and write rates of 4854MB/s and 6440.3MB/s respectively, 9to5mac discovered that a new M2 Pro MacBook Pro with 512GB storage is notably slower, reporting read and write speeds of 2973.4MB/s and 3154.5MB/s (see: 9to5mac.com/2023/01/24/macbook-pro-ssd-performance-drop).

The reason for these discrepancies in performance comes down to the number of NAND modules used to implement the storage capacity. For example, in the case of the M1 Mac mini with 256GB of storage, the 256GB was configured with two 128GB NAND modules, while the M2 Mac mini with 256GB of storage implements this same volume with just a single NAND. And the reason two NANDs are better than one is because, simply put, much like a RAID 0 array where data is striped across multiple disks, you can achieve faster access times by employing the services of multiple disks simultaneously.

In terms of audio streaming performance (for recording and playing back audio tracks, feeding samplers, and so on), these differences in storage speeds probably don't matter all that much. Even a speed of 1500MB/s still allows for around 2700 channels of 192kHz audio as a theoretical maximum. However, the significance of storage performance also depends on how much unified memory you have in your Mac. With an 8GB system, for example, Mac OS will rely more heavily on virtual memory, paging blocks of memory to and from the internal SSD more often, in order to make the most of the physical memory available. Therefore, the more demanding your workload — which will be determined by the applications you need to run and the size of the sessions or projects or songs that you're working with, including large sample-based instruments — the more memory you'll need to access to keep the system running smoothly.

This essentially means that Macs with SSDs comprising multiple NANDS will perform better under heavier workloads. So, if there's one specification that's important to account for, it's internal storage, especially when using your Mac for creative professional purposes, such as music and audio (and video).

And talking of audio and video, there are many factors that affect how general performance, as measured using tools like Geekbench, translates into the ability to run lots of plug-ins or bounce huge mixes in seconds. So, next month, I'll be doing a deep dive into the performance of these M2 Macs with Apple Silicon-compatible music programs, including Logic Pro X, Pro Tools and Cubase.

$ Mac mini from $599; MacBook Pro from $1299.
W www.apple.com

Studio Vs Mini

Although the M2-based Mac mini is clearly meant to be an affordable, entry-level system that still punches above its weight, the more expensive variant with an M2 Pro seems, at least on the surface, to be positioned perfectly in terms of price and performance with respect to the Mac Studio.

Starting at $1299 when ordered with a 10-core M2 Pro featuring 16GB memory and 512GB storage, this is $700 cheaper than the base model Mac Studio with an M1 Max, 32GB memory and 512GB storage. Even if you configure your Mac mini with 32GB memory and 1TB storage for an additional $400 and $200 respectively, you'll still save $100, unless you require the faster 10 Gigabit Ethernet option. However, once you match the Mac Studio in terms of price, it probably makes sense to opt for the Mac Studio for its greater graphics and video capabilities, and front-panel connectivity (including an SD card slot).

On the other hand, if you opt for the Mac mini with a 12-core M2 Pro (which offers roughly a 16 percent improvement in CPU performance over the M1 Max), stick with 16GB of unified memory and Gigabit Ethernet, but upgrade to 1TB of storage, this configuration will cost $1799 — $200 cheaper than the Mac Studio. And this might make sense, especially if you don't require more powerful graphics and video processing.

These comparisons will inevitably change depending on what Apple do with the Mac Studio going forward. For example, if an upgraded Mac Studio is released with the same price and configuration as the current model, but starting with an M2 Max, this would make higher-end, more expensive Mac mini configurations less desirable.

Sonicware SmplTrek

Sampling Groovebox

Sonicware's SmplTrek packs an ambitious sampler, sequencer and recorder into a single portable box.

ROBIN VINCENT

The Sonicware SmplTrek may have lost a couple of vowels in its fight for position amongst other portable sampling grooveboxes, but with integrated MIDI sequencing, live looping and mixing, it might well have a lot more to offer.

The workflow feels a bit like a hardware version of Ableton Live Lite, or perhaps more accurately like Roland Zenbeats with its orientation of tracks at the side and scenes along the top. In full flow, it's really more of a hybrid digital portastudio than a beat-making groovebox, as it combines live recording with clip launching, scene triggering and playful sampling endeavours. Finding the flow can take a bit of patience and dedication as you first have to wrestle the interface into submission.

Form

SmplTrek is a thick but compact box of black plastic designed in part to be held in two hands like a portable games console. As you grab either side the navigation D-pad and a couple of buttons locate right under your thumbs, although I found that there's not much you can do with just those controls, so it inevitably ends up on your desk. It has a handy screw thread on the bottom for mounting it on a mic stand for a more live-performance vibe.

Across the front, you have 16 scene buttons and 15 track buttons, which double up as velocity sensitive

Sonicware SmplTrek
$429

PROS

- Different tracks types.
- Three audio tracks with take selection.
- Easy sampling to drum kits.
- Clip and scene launching.
- Sample from anywhere.
- Can also be an audio/MIDI interface.

CONS

- Quirky interface.
- Small screen.
- Fiddly knobs.
- Small pads.
- Underwhelming library.

SUMMARY

SmplTrek could be your sequencing, sampling, looping and recording songwriting partner, where the versatility balances out a sometimes challenging interface.

finger-drumming pads. There are a handful of other buttons to get to the things you need, and everything lights up beautifully except for the array of black, Volca-style knobs, which are really hard to see. In the centre is a small but readable screen about the size of an Apple watch; this is where the action happens.

On the back is a pair of quarter-inch line inputs that can handle guitar and microphone levels, and a pair of line-level outputs. Very welcome 5-pin DIN MIDI in and out and a DC power socket complete the back panel. On one side, you've got a micro USB port and SD card slot. SmplTrek has no internal memory, so everything's done on the SD card (not included). The USB port is useful for data transfer but also turns the SmplTrek into a 2x2 audio/MIDI interface and controller for a phone or computer. On the front edge there's a 3.5mm headphone socket, and on the top we've got 3.5mm analogue sync in and out.

In line with Sonicware's love of portability, the SmplTrek can take six AA batteries, although it did come with a power supply which, I think, is a first for Sonicware boxes and much appreciated. It has a reasonable built-in speaker so you can impress your friends with your beats while out and about. For sketching out ideas or sampling on the go the built-in mic is brilliant, but you'll need a pair of headphones if you don't want to risk feedback or find yourself resampling the output of the speaker.

Overall, it's quite a neat box, with a good weighty feel and a clean layout that doesn't look too complex or busy. And that alone makes you feel you can pick it up and go and get on with making some music.

Function

The SmplTrek has five track types to choose from, which dictate the content of the track's clips. You can set a track up as an audio loop, or a one-shot sample; you can have a sample-based instrument, a drum kit track or a MIDI track. There are no virtual or hardware instruments, so everything is sample-based unless you MIDI out to an external synthesizer.

The review model was running firmware 1.4.99. This brought in some very welcome expanded features and improved some of the more laborious user interface functions. The interface can be very challenging. It's almost like a not-so-fun side-quest in a video game designed to distract you from your ultimate purpose. You just have to grit your teeth and learn your way around it, and it does start to open up.

The SmplTrek has 10 tracks for your sampling, drumming and sequencing clips that you can arrange over 16 scenes. The latest updates boosted the number of drum and instrument tracks to four of each. In addition, there are three audio tracks for recording vocals or live instruments alongside your song. You can also route through an external source or connect a computer or phone via USB for sampling or mixing alongside. To help with the mixing, there's EQ and a choice of 32 effects.

Loop Mode

The original concept for the SmplTrek seemed to be to keep the sampling and sequencing as two different functions. You would spend some time capturing and processing samples and then use the sequencer to import them as loops or instruments for arranging. This created a bit »

The SmplTrek's back panel offers quarter-inch audio I/O and, joy of joys, full-size MIDI I/O ports.

» of a disconnect as you couldn't sample into the sequencer, so you would have to keep navigating to and from the sample pool to find your newly sampled samples and bring them back in. With recent firmware updates you can now sample directly into a clip.

Being able to record directly into a clip is what gives SmplTrek its live looper feature. It's not quite what I expect live looping to be. Rather than automatically overdubbing, you have to hit the right combination of buttons to record another loop alongside the first on another track. If you set up each track to be in loop mode then you can step down through them quite smoothly in order to keep layering up loops. But it doesn't have the flow of features, the overdub, undo/redo and so on that you'd expect to find in a dedicated looper.

You can transfer your own loops and samples to the SD card and import them into the clip, which you should undoubtedly do because the included library is almost entirely non-existent; not a drum loop in sight. In the Preset folder, you get about 24 single instrument samples that are intended for instrument tracks, and 12 drum kits. You can download a 1GB sample pack free from a third party, which is not very easy to navigate and has a naming convention and folder system that doesn't lend itself to the tiny SmplTrek screen. But you could see it as an encouragement to build your own library. When importing you just have to be aware that there's no automatic time-stretching or beat detection, so you'll need to do some editing to get your loops to fit.

One Shot

For a one-shot track, you are pushed directly into the pool where you can explore the sample library, such as it is. As with the loop mode you can, of course, load your own samples, or sample something straight in, but unlike loop mode this happens outside of the track, in the separate sampling section. If that sounds a little confusing, then that's a feeling you get used to when learning the SmplTrek.

Anyway, once you've selected your sample from the pool, you can leave it to be triggered on the pad in a live performance or sequenced in a pattern. These are not melodic tracks as such; this is more about cymbals, percussion and sound effects being triggered, but you can do all sorts of things in the automation lanes. You can edit the pitch, velocity, level, panning, send amount, micro-tuning and gate length for each step in the pattern. The editing detail is really very good, as is the sample editing, which we'll come to in a minute.

"Once you see it as a songwriting tool, as a portastudio with bells on, it becomes a uniquely powerful and portable device."

Instrument

Instrument and MIDI tracks essentially work in the same way, with a piano-roll style sequencing view. For MIDI, each new track gets an automatically allocated and incremental MIDI channel number, whereas for instruments, you can select from any of the 24 now-legendary sample sounds or your own single-note samples. The track pads now turn into a little one-octave keyboard that you can shimmy up and down in pitch with some side buttons.

You can record your sequence in real time or step time in up to eight notes of polyphony. If you don't fancy the little pads then you can connect a MIDI keyboard for some more musical leeway. Step recording seems to offer every way you can think of to input notes. You can thumb around with the D-Pad, find the note and position you're after and press OK to enter the note. Or you can find the note and press the step number button to enter a note at that step, or get the cursor to the step and press the pads to enter the pitch. Entering and editing notes can only be done while the Record button is active.

You have similar automation options as with the one-shot samples, but you will have to thumb around for the individual notes in order to apply them. You can also add an attack-release envelope to the overall sound. As with the other tracks, the level of editing detail is nice, if a little time-consuming with the D-pad and tiny screen. The knobs are not very speedy either. They are clicky, and it takes a lot of winding to move through 128 values, no matter how fast you spin them.

Drum Kits

The SmplTrek comes with 12 drum kits covering all the usual suspects. They »

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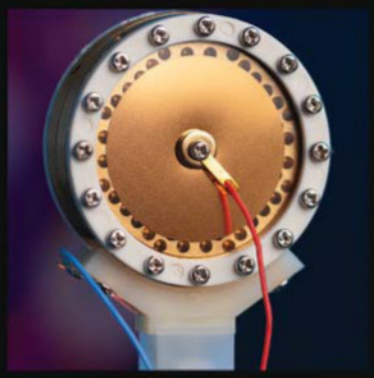

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» are single-sample sets that fill all 15 of the velocity-sensitive pads. You can get some expressive playing if you treat them gently, but even mild finger drumming tends to push them to the max. And these pads are small, smaller than the end of any of my fingers, so finger-rolls are tricky and accidental triggering of neighbouring pads happens quite a bit. However, their size and proximity make it easy to access all 15 pads under the natural fall of your fingers, so, in the context of the device, I feel the pads work really well.

There are a number of editing pages available for a drum kit. For the kit as a whole, you have level and a 'LoFi' option, which possibly removes a little bit of top end, but it's really hard to tell. If you dig down into the 'Instrument Parameters', you'll find individual level and panning controls for each pad, along with effect send, pitch, reverse and slope (envelope). You can have up to six choke groups for managing hi-hats or keeping a monophonic feel to certain hits.

You can build your own kits from imported samples by adding them to pads, or you can sample something in, chop it up and throw it into a kit. When I first got started with the SmplTrek the process of creating your own kit from sampling was so convoluted that it didn't feel like a good use of my time. It's now been smoothed into a flow that goes something like: sample it, save it, find it, chop it, pad it, load it and off you go. It's no trouble at all.

Drum patterns work like the instrument track piano roll. You can record in real-time, or use the D-pad, step buttons and pads to set out your pattern manually. Quantisation is always present from 1 to 32 notes per bar, which is great for keeping in time, and you can enable a metronome and set a pre-roll.

Sample Editing

Sampling into SmplTrek tends to be a separate process which you can get into at any time by pressing Func+Record. You can select any of the inputs, including internal mic, external mic, guitar, line input stereo or mono, USB input, or capture the output of an existing track. There's also an unexpected guitar tuner, which is a nice bonus.

In the sampling page, you can check your level and hit Start to start recording, and your audio is painted on to the screen. You can then play the sample back, zoom in and scroll around it but not much else. To take things further, you need to save it and then re-open it from the sample pool.

The SmplTrek's 'Global' mode is actually a three-track recorder that works alongside your sampled patterns.

When loading a sample from the pool you have some decisions to make from a menu. You can load it into a clip as a loop, one-shot or instrument track. You can chop it up manually or with auto-slicing to generate new samples or a drum kit. You can opt to trim the sample by setting a start point and using bpm and number of bars to set the size, or just dial in a start/end point. Time-stretch is another option, either melodic or rhythmic, which again works on bpm and bars. Other options include normalisation and changing the mono mode.

There's no beat detection or real-time time-stretching, so to make things fit together, you have to know the bpm of your material and process it accordingly. This is probably the only weak area in an otherwise decent selection of sample editing tools. It would be fabulous to be able to drop in your tempo-infused loops and have them automatically lock into the project tempo, but doing it by hand is not too taxing.

Arranging

I mentioned how the songwriting workflow of SmplTrek is a bit like Ableton Live or Roland Zenbeats. What's important to know is that the Scene dictates what's in play. You can't launch clips or one-shots willy-nilly; you can only launch what's in the current scene. So from the SmplTrek's orientation this means one column of 10 clips filling the 10 tracks. Each clip can have its own length and number of loops or be empty and ready for live playing, but the length of the scene ultimately takes precedence.

If you press Play, your whole project will play back left to right taking into account any scene repeats you've set up to create your song. In launch mode you can either launch a whole scene by pressing the corresponding button or select a scene and launch individual clips within it. Once it's playing, I couldn't fathom a way to move to another scene without launching it and all its clips; I could only do it while stopped. So the scenes restrict what you can do to groups of 10 clips, rather than freeform access to a potential of 160 from purely a clip launcher point of view.

You can copy/paste clips and scenes to help you fill out your project, and of course, each clip can have its own pattern or sequence playing the selected source of the track. There's no indication on the screen to say whether the clips are the same or different, so you've got to keep your wits about you. It's relatively easy to navigate around the clip grid, program in your patterns, rearrange the clips and work towards an arrangement.

Global Tracks

The oddly named 'Global' tracks are three audio tracks that run alongside your project. You won't find them in launch mode, but when you play the whole project back these open up for you to record a full vocal performance, a guitar track, harmonies or whatever you want. This is a huge feature that should

probably take more of our attention. While you can record yourself into clips as samples and arrange it in scenes as a vocal line, the ability to record yourself singing along to your project transforms the SmplTrek from what could be seen as a looper or clip arranger into a portastudio. And you have three tracks, and each track has three takes that you can freely swap between.

The Global tracks are tracks 11, 12 and 13 in the project, and when you go into them they are presented on a timeline of bars and corresponding scenes. Once you've recorded your three takes, you can split up the audio to swap between them or mute sections. You can record in from any of the usual inputs, which means you could also bounce one or more of the other tracks to a Global track to free those up for something else.

I think the sampling and looping side of SmplTrek is a huge part of its identity, but it's all too easy to be fooled into thinking it's just a fun sampling box. If you refocus on the Global tracks then SmplTrek becomes a three-track studio with a built-in drum machine, sequencer and sample looper. That's a very different vibe and would open it up to people who might otherwise assume it's not their thing.

Mixing & Effects

Finally, we have the mixing section, where we can combine and process the 10 tracks, the three Global tracks and the USB and external audio. With level metering, you can see up to eight tracks on the screen at once. Each channel has level, panning, send and some EQ filtering that's light on the EQ and heavy on the filtering.

There are plenty of effects inside the SmplTrek, but the allocation is a little disappointing. There are insert effects, send effects and master effects, but you can only have one of each at one time. You can load up a reverb for the send effect and send all the tracks to it, so that's not too bad. For master effects, it's really an overall lo-fi or compression option, which are fine. But for the insert effect this feels very restricted. However, there isn't endless amounts of processing power in the SmplTrek, so you've got to think about it regarding resampling and signal chain. If you want to add an effect to something, you could resample it back in with the effect applied and free up the one effect for use elsewhere. Or you can put the insert effect on an input and record in wet. There are bass amps and guitar amps in here expressly for that purpose. So in that respect, the range of effects on offer and how you can use them is pretty comprehensive.

Conclusions

Comparisons to the Roland SP-404 MkII seem obvious. When the SmplTrek was originally released as a Kickstarter, the price undercut the SP-404 by a couple of hundred dollars and so it was described as a potential 404-killer. But it's not an accurate or helpful comparison. The SP-404 MkII is the master of lo-fi beats and a sample-triggering performance instrument; the SmplTrek is a songwriting machine with audio tracks, sampled beats, MIDI sequencing and live looping. Now the SmplTrek is on the open market, the prices are much more similar and it has to work harder to assert itself.

Once you see it as a songwriting tool, as a portastudio with bells on, it becomes a uniquely powerful and portable device. While it can make beats, we need to think beyond that urban sampling stereotype and picture it on a desk next to a guitarist, a vocalist, a live streaming performer, a DAWless producer, songwriter and sound designer.

There are still some quirks and avenues of frustration in the interface that are difficult to ignore. I'd like an undo function more than anything. At times it felt like I was being hampered at every turn. Sonicware have come a long way with the firmware in only a few months, and the improvements are a source of relief; keep them coming. There are also none of the more esoteric functions like randomisation, probability, rolls or ratchets, or an easy way to reach in for some live performance filter or effects action.

However, the SmplTrek is a great alternative to a computer-based home studio and far more portable with a lot less fiddling about. It's not as intense as an Elektron box or as deep as a Roland, but it does a wider range of tasks well enough to give you a solid songwriting partner.

$ $429
W www.sonicware.jp

Chandler RS660

Valve Compressor

Can't decide if you'd rather blow your vari-mu budget on a Fairchild or an RS124? Chandler's RS660 puts a little of both in one box...

BOB THOMAS

Chandler Limited's exclusive agreement with EMI/Abbey Road Studios has enabled founder Wade Goeke to bring to market not only accurate recreations of some of the most revered studio equipment of the 1960s, but also to develop new products that are inspired by those classics. Alongside the various legendary EMI preamps, EQs and compressors, for example, Chandler's range includes original and EMI-inspired rackmount designs and 500-series modules, as well as two innovative takes on the classic large-diaphragm capacitor microphone, in the form of the valve-based REDD and the solid-state TG.

Design & Construction

Their latest product, the RS660 compressor, is a hand-built, transformer-balanced, single-channel, variable-mu valve compressor-limiter. Despite its decidedly retro appearance, though, it's arguably as much a child of the 2020s as it is of the 1960s. Their first valve compressor, the RS124, was essentially a painstaking recreation (albeit a lightly enhanced one) of EMI's 'reworked' Altec 436B, whereas this new model is an elegantly conceived fusion, combining qualities of the RS124 with others from its famous contemporary the Fairchild Model 660. It also tips its hat slightly in the direction of the EMI TG1 diode-based limiter, which was itself originally designed to emulate the operation of a Model 660.

The dark brown fascia is bounded by a black trim plate that carries the unit's official designation plus the Chandler, EMI and Abbey Road Studios logos, and also defines the limits of the control area. Within those confines are a momentary push switch and associated potentiometer that enable you to compensate for any asymmetry in the unit's push-pull output. There's also a large, square, mechanical meter that displays compression on a scale of 0-30 dB, an input level control (scaled simply 1-10), an output level control (-10 to 0), a three-position switch to set the operating mode, and a switch to choose between the seven preset attack and release time combinations.

The control knobs are made of plastic rather than 'period correct' Bakelite, but nonetheless combine with the power switch and its red-jewelled power-on light to give the fascia a distinctively vintage vibe very. A slightly more modern-looking toggle switch activates the side-chain link between two connected RS660s.

The rear panel carries the XLR connectors for the transformer-balanced input and output, a TS quarter-inch jack socket that carries the side-chain link for stereo operation, and a fused IEC mains input. A less familiar sight, unless you happen to own an RS124, is the switch that selects between either 200Ω (Abbey Road's standard back in the day) or 600Ω output impedances. The input impedance is fixed at 15kΩ.

Taking the top off the RS660 reveals an internal transformer, valve and circuit board layout that, at first glance, bears an uncanny resemblance to that of the RS124. But closer inspection reveals that the circuit and power supply boards are somewhat more complex. Component quality and the level of attention to detail in the hand-wired assembly are all of the exemplary quality you'd expect to find

While it's primarily a compressor-limiter, the RS660 also features a switchable THD mode, which allows you to add more grit and warmth to a signal.

in a unit costing what the RS660 does. As with the RS124, this compressor is configured around a simple three-valve, transformer-balanced, differential signal path derived from the Altec 436B. In the RS660, the variable gain stage comes in the shape of a single 6386, rather than utilising the 6BC8 found in both the Altec and the RS124. It's the same type as that used (in a series-wired multiple of four) in the Fairchild Model 660. The substitution of the 6BC8 by the 6386 is the only valve change, with the RS660's side-chain and push-pull output stage being built, respectively, around the same combination of 6AL5 dual-diode and 6CG7 dual-triode as found in the 436B and RS124.

In order to allow the correction of any drift in the balance of its push-pull output stage and the audible low-level modulation that resulted, the RS660 features EMI engineer Len Page's ingenious method of balancing the two sides of the output stage. It's something he originally developed for the Altec 436, to avoid having to stop a session while the unit went back to Abbey Road's service department for realignment. In this elegant solution, which inhabits the RS660's front-panel balance control section, depressing the momentary button switch feeds a series of clicks through the unit, and these are nulled out using a slot-headed potentiometer. Once nulled out as far as possible (the clicks never fully disappear) the output stage is balanced.

In a nice homage that's carried over from Chandler's RS124, the white-lettered front panel 'BAL' legend for this control is in the same typeface as the label stuck onto the front panel of the original EMI-modified Altecs, and it will be familiar to any user of Dymo labels!

Controls & Modes

The input and output controls' potentiometers on the standard version of this compressor are both undetented, and this, in combination with the rather coarse 0-10 front panel scaling, means that you'll have to resort either to photographs or an old-school Chinagraph pencil for accurate recall. However, as a cost option, the unit can be bought fitted with two stepped switches, which is obviously more convenient in that respect.

There are three switch-selectable operating modes, called Limit, Comp and, somewhat unusually, THD (Total Harmonic Distortion). The last is a signal artefact that manufacturers have traditionally worked hard to minimise, but in recent years it's become desirable to have more control over the amount of 'warmth' imparted by studio processors. The THD mode is inspired by a circuit used in Chandler's TG1 limiter, which employs a diode network. When the RS660's mode selector switch is set to THD, the unit's compression and limiting function is disabled (as in the TG1), which means that, since all its transformers and valves remain in circuit, the RS660 has the ability (again, like the TG1) to add overdrive and harmonic distortion to any signal passing through it.

Just as you'd expect, selecting Limit or Comp switches the RS660 into its limiter or compressor modes, respectively. As a limiter, the RS660 has been deliberately designed to be "fast, coloured and lively, with the ability to add excitement to any source". Switching from limiter to compressor mode raises the threshold, lowers the onset of compression, makes the knee smoother and less aggressive and modifies the gain structure, which makes for a more controlled, less aggressive response overall. »

Chandler RS660
$3399

PROS
- A wonderful sound, full of vintage tube character.
- Can impart warmth, saturation and presence.
- A very usable range of attack/release time combinations.
- Extremely intuitive.

CONS
- No bypass function.
- Side-chain filter would be nice.

SUMMARY
A valve-based compressor that builds on the heritage of the two compressors made famous by Abbey Road Studios in the 1960s to deliver a superb audio performance full of vintage character, warmth and tone.

The RS660 features a switchable output impedance, and stereo linking courtesy of a TS jack on the rear panel.

» The seven-position Time Constant switch sets the RS660's attack and release time presets, with position 1 being the fastest and 7 the slowest. The first three positions are designed to replicate the coloured, faster settings of the Fairchild Model 660, while the fourth position has a more moderate, less coloured speed — it's a halfway house between the faster settings and "purposely quite slow, and influenced by the gentle, uncoloured and low-artefact sound of many vintage valve compressors, including the EMI RS124." Chandler don't provide more detailed specifications for any of the RS660's attack/release settings, but although my inner geek might well feel better for knowing them, that lack of knowledge really doesn't matter in the real world. It's incredibly easy to get the sound that you want from the RS660 — provided, of course, that you enjoy its vintage vibe.

As I've mentioned in a previous review, the current reissues of the 6386 valve do not perform quite like their venerable forefathers; they exhibit lower gain, less compression and significant tonal differences. This has led some designers to replace the reissue twin-triode 6386 with a pair of 6BA6 pentodes wired as triodes. Instead, Wade Goeke has invested time and effort in ensuring that the new 6386 valve sounds like the vintage version, and judged on the sonic character of the RS660 he has succeeded in that.

In Use

The RS660 produces superb results across a wide range of compression, from the subtlety of a few decibels of gentle compression to needle-pinning heavy limiting. With only an input level, seven time constants and an output level to play with, it's much like the Fairlight 660, in the sense that it's very much an 'experiment and get to know it' sort of processor. Once you've developed a sense of how the RS660 responds, though, you'll find that dialling in the feel that you want in a track is a quick, intuitive and natural-feeling process. Having said that, while its sound and compression characteristics mean it can excel across a channel or bus insert, it does lack a hard bypass facility, making in/out comparisons tricky if the insert point itself can't be bypassed easily. Using it as a parallel compressor makes such comparison easier and opens up the opportunity to high-pass or otherwise EQ the input to the compressor.

"Wade Goeke has invested time and effort in ensuring that the new 6386 valve sounds like the vintage version."

As with many other high-end, valve-based compressor-limiters, the RS660 has that uncanny ability to add warmth, character and presence to any sources running through it, and it does that particularly well for vocals, acoustic guitars, bass and drums. But I also found the separate THD function (which is more about warmth and character than the sort of full-on distortion you'd look for in a pedal) very useful for adding a bit of grit and harmonic colour to rocky vocals, drum tracks and DI'ed bass in particular.

With only a single RS660 available to me during the review period, I wasn't able to fully evaluate the performance of a linked pair for stereo use, but I have absolutely no reason to think that there would be any problem in that regard, providing that you have the two units set to identical input levels and time constants. On that note, if its performance in a stereo bus/mastering role is important to you, it's worth pointing out that there's an option to purchase a matched pair with stepped input and output controls.

Verdict

With the RS660, Wade Goeke has successfully combined the essential qualities of the Fairchild 660 and the EMI RS124. The result hints at a development approach that, given the opportunity, EMI's engineers might well have taken back in the day. This is a marvellous compressor-limiter whose audio and operational performance and versatility make it an extremely attractive alternative to the usual suspects at the premium end of the compressor-limiter market — it more than justifies its professional price tag, and I'm going to be very sad to see this review unit leave my studio!

$ Standard model, as reviewed $3399. Stepped control version $3599 each or $7550 for a matched pair.
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Universal Audio Galaxy '74, Del-Verb & Max

Tape Echo, Reverb/Delay & Preamp/Compressor Pedals

The latest additions to Universal Audio's UAFX digital stompbox range offer faithful recreations of classic studio tape delays, reverbs, compressors — and even a vintage microphone preamp.

DAVE LOCKWOOD

When Universal Audio (UA) launched their range of digital, compact pedals with the Starlight delay, Golden reverb and Astra modulation processors in 2021 (reviewed in *SOS* July 2021: www.soundonsound.com/reviews/universal-audio-uafx), their audio quality was never in doubt. But the absence of MIDI control and very limited preset handling was certainly viewed as a compromising factor for live performance when compared with some competing products. The second set of UAFX launches was another trio, this time of amp and speaker simulations of specific classic Fender and Vox models (*SOS* July 2022: www.soundonsound.com/reviews/universal-audio-uafx-amp-pedals). Despite, in my opinion, setting a new benchmark for realistic

Universal Audio Galaxy '74, Del-Verb & Max
$349 each

PROS
- Great-sounding, very different compressors in the Max.
- Del-Verb offers the stellar sounds of Starlight and Golden in one box.
- Galaxy: it's like an old RE-201 that you don't have wonder if it's still going to work every time you turn it on!
- The analogue emulations are all just startlingly real and characterful.
- The Dyna Comp isn't noisy, but everything else is still there!

CONS
- No dedicated reverb decay time control in the Del-Verb.
- No MIDI or real-time external control.

SUMMARY
These may not be what most of us were expecting. They may not even be what we wanted — I wanted a Marshall, a Mesa and a line-level OX! — but for all that they are not, they are undeniably superb-sounding pedals that will surely inspire and enhance anyone's music making.

digital modelling of the sound and feel of a miked-up guitar amp and speaker, the continued absence of MIDI control and limited preset handling was still disappointing to some. Perhaps the only things that everyone agreed on, however, were that the 'glaring omission' from the amplifier line-up — a classic Marshall or two — was bound to be addressed in the next round of releases, and that the somewhat underwhelming app would soon get a significant upgrade into a proper software editor.

Well, we weren't right. The latest clutch of pedals, comprising the Galaxy '74 Tape Echo & Reverb, Del-Verb Ambience Companion, and Max Preamp & Dual Compressor, are still without MIDI control or expanded preset handling, and that managed to send certain corners of the Internet into 'maximum outrage'. The Max and Galaxy pedals actually have no presets at all, just functional settings within the app, while the Del-Verb has a choice of fundamental 'voicings' that you can assign to the different delay and reverb stages of the pedal.

I know there are some people who feel that a digital pedal at this price point simply must have MIDI control as an option, but to me, each of these releases is improved by its lesser dependence on the app for normal functioning. The app now does 'choose and use' stuff in these units, with everything else under your fingers for immediate tweaking. Not so much under your feet as some would like, perhaps: just bypass, tap tempo, on/off for separate stages, and a bit of wacky feedback runaway in the Galaxy. The frustration in some of the earlier UA pedals lay in the fact that they could do so much, whilst making so little of it instantly accessible in a live performance context. These models are perhaps a little less broad in their scope, but the end result is something that I find to be more focused.

Hardware

These pedals are all housed in the now-familiar UAFX dual-footswitch metal casing, with two banks of three rotary controls, two or three mini-switches, and no displays or readouts beyond switch-associated LEDs. Bypass switching is buffered, seemingly instantaneous and noise-free, although selecting a new virtual device (eg. a different compressor or delay type) does cause a brief audio interrupt. Two pairs of unbalanced TS jacks offer stereo audio input and output. A stereo input pair obviously facilitates maintaining any stereo element in the input source, but all UAFX pedals also run dual instances of their processes, so delays and verbs that have no real stereo element to them audibly seem to have some kind of subtle 'imaging' going on when you run them in stereo. Of course, if you want the most authentic experience of a Roland Space Echo from the Galaxy '74 or the modelled MXR Dyna Comp in the Max, you'd run it in proper mono!

Powering, using the standard 9V DC centre-negative format, needs to come from an external, isolated source with at least 400mA current capacity; there is no battery option and a PSU is not included in the purchase price. The remaining connection is a USB-C socket, used only for firmware updates via the UA Connect desktop app. Pedal settings have to be done via a separate UAFX Control mobile app over Bluetooth. The Bluetooth 'connection experience' was faultless this time, although I and others had a few issues with the earlier units. Still, I fail to see why these settings can't also be made available in the desktop app for anyone who is having trouble connecting, or just as a backup. Although clearly optimised for instrument-level connection, they have headroom up to +12dBu and I've never had »

All the UAFX pedals run sync'ed dual instances of their processes, so there is often some subtle stereo imaging even in nominally mono effects.

» a problem using these with keyboards or nominally line-level mixer sends. You won't clip the unit with anything sensible, though you may affect the way the processing responds.

Del-Verb and Galaxy have a zero-latency, analogue dry path through them, but there's about 3ms latency in the Max unit because it processes the 'dry' signal like the amp sims, which have circa 2ms. With no digital interconnect between them this is cumulative when you use them together, but in no normal application in which I have used these has this been a problem. I can't feel anything amiss with a Max and a Dream or Ruby in series — you feel it long before you can hear it — but all latency is cumulative, so you might have to think about it if you are going to be passing a chain of these through other processes with further latency of their own.

To The Max!

I must admit, when I first learned about these units, the Max Preamp & Dual Compressor is the one I didn't really understand. It has two 'channels', each offering a choice of one of three digitally modelled comp/limiters, namely UA's 1176, a Teletronix LA-2A and an MXR Dyna Comp pedal, as well as UA's renowned 610 tube preamp. The channels can be used independently or together, in series or parallel. For anyone not familiar with these all-time classics of the compression world, the 1176 is a FET-based design, offering relatively clean and precise compression, whilst the LA-2A is both tube-based and uses an optical gain-reduction element, making it warmer and more subtle, if less controllable. The MXR Dyna Comp might not be a studio hardware classic, but certainly deserves its place in history as the first really usable guitar pedal compressor. It's noisy, dirty, lo-fi compression with lots of side-effects and almost useless with humbuckers — yet, it's an effect that I and countless other guitar players still have a use for!

A Dyna Comp, with its fixed ratio and attack and release times, distortion and noise, is something you'd only use in front of an amp, processing a raw guitar signal, in the real world at least. The 1176 and LA-2A are line-level devices that you'd normally use on an insert point or after a mic pre, not into the front of a guitar amp. So if I'm using a Dyna Comp, why would I want to shove some more level into it via the 610 mic pre? It goes into shock when I hit it with a high-output pickup already! But, hang on. This *isn't* the real world. This is the world of digital modelling, where even 'authentic emulations' can still do 'new things'. Just five minutes into using it I was taking back my initial reservations: Max is an absolute gem! Fantastic for slide/bottleneck guitar. Brilliant for just putting some 'bounce' under the fingers when working at low volume.

Forget conventional usage, this works brilliantly in any application, shaping signals, subtly or overtly, whether in front of an amp or amp sim, or just DI'ed, or on a mixer insert point. I really got into using combinations of compressors in the safe world of plug-ins. It is more of a challenge in the hardware domain, of course, dealing with appropriate gain-staging and the inevitable rising noise floor. None of that is a problem here: you're free to create stacked or parallel combinations of different compression characteristics for maximum squeeze, ultimate subtlety and anything in between. You still have to be aware of levels to some extent — you can easily take the Dyna Comp outside of its useful working range — but the output control setting is unique to each compressor, so you can balance out a stacked LA-2A into an 1176 for a bass DI, for example. The only real omission here at present is an I/O configuration that allows you to place the Dyna Comp in front of an amp or sim, and the other compressor stage on the output, but I understand that an extra routing mode called Split (Dual Mono) mode will soon be added via a firmware update, which will let us do exactly that!

There isn't a dedicated dry-path 'blend' control for a parallel compression setup but, of course, if you set the comps in Parallel mode, then set the first compressor's Comp setting at minimum, that becomes a dry path that you can balance against a smashed signal from the second compressor stage.

Staying In Control

The control settings are independent for each compressor channel, but there is only one set of physical controls. So there will be 'what you see is *not* what you get' moments. When either one of the compressor channels is selected on its own, the controls are all active on that channel. When both channels are activated simultaneously, the knobs are controlling either compressor 1 or 2 according to the position of the Comp Select switch. Logical enough, but when you switch between channels, you get the settings that you last made there, even though the knobs themselves may now be in completely different positions. As soon as you move one, it becomes active and you'll hear the sound 'catch

up' to the new position. Sensibly, sounds seem to slightly 'ramp' to new positions, reducing the chances of giving yourself a big audible surprise. By default, the Max operates in series mode, in which both channels can be active at the same time, with Comp 1 feeding into Comp 2. Both can be on in parallel mode too, with the input signal feeding each comp separately before recombining at the output. There is also a toggle mode, Exclusive, in which either footswitch flips from the current compressor to the other one, allowing you to perhaps set up a rhythm sound with mid-scoop on Comp 1 (with the 610 set to LF/HF boost) and use the second compressor stage as a solo boost with a bit of a mid hump (with the 610 set to LF/HF cut).

There may be only one control knob, but there are actually two 610 mic preamps on board, with independent settings available for each. Set fully anti-clockwise, the preamp has unity gain, with its EQ (adjustable only within the UAFX Control mobile app) set to off. If you want to use the EQ without adding any gain, you can nudge the control just above off and you'll have access to a low shelf EQ that operates on frequencies below 100Hz, cutting or boosting by up to 6dB. A high-shelving stage similarly addresses frequencies above 10kHz.

Turning up the preamp control progressively adds a bit of classic 610 'tube flavour' and additional harmonics, as well as allowing you to push the front end of the compressors a little more... or a lot!

The app settings also include a useful side-chain filter for the 1176 that removes some low-frequency content from the compressor's control signal, reducing the potential for overreaction to the high energy inherent in low-frequency signals that might generate audible 'pumping' in the audio output.

The Comp control sets the compression amount, but in a different way for each compressor, replicating the effect of the Sensitivity control on a Dyna Comp (turn up for more squash and sustain) and working like the Peak Reduction knob on an LA-2A or the Input knob on an 1176, so you turn up to increase compression by lowering the threshold. Make-up gain is applied automatically in the LA-2A to compensate for level lost through compression, with independent final output level for each compressor stage available via the Output control.

An original hardware LA-2A only has two ratios, of course, but no-one is going to complain about having access to a few more: 4:1, 8:1, 12:1, 20:1 and 'All'. The 4:1 setting apparently matches the original Comp setting and 20:1 replicates the Limit setting. The 1176 also includes the much-loved 'all ratio buttons selected at the same time' setting. Similarly, a real LA-2A has no release control but the »

ALTERNATIVES

You can't go wrong with premium products from the likes of **Eventide**, **Strymon** and other manufacturers of a similar calibre, but I'd hesitate to point to specific models as competing alternatives when there is an obvious sound/facilities balance to be struck. We are lucky to live in an age where we are faced with such choices!

HP-1
Personal In-Ear Amplifier
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DJ PREII
Phono Pre-Amp
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-Popular Science

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MX622BT
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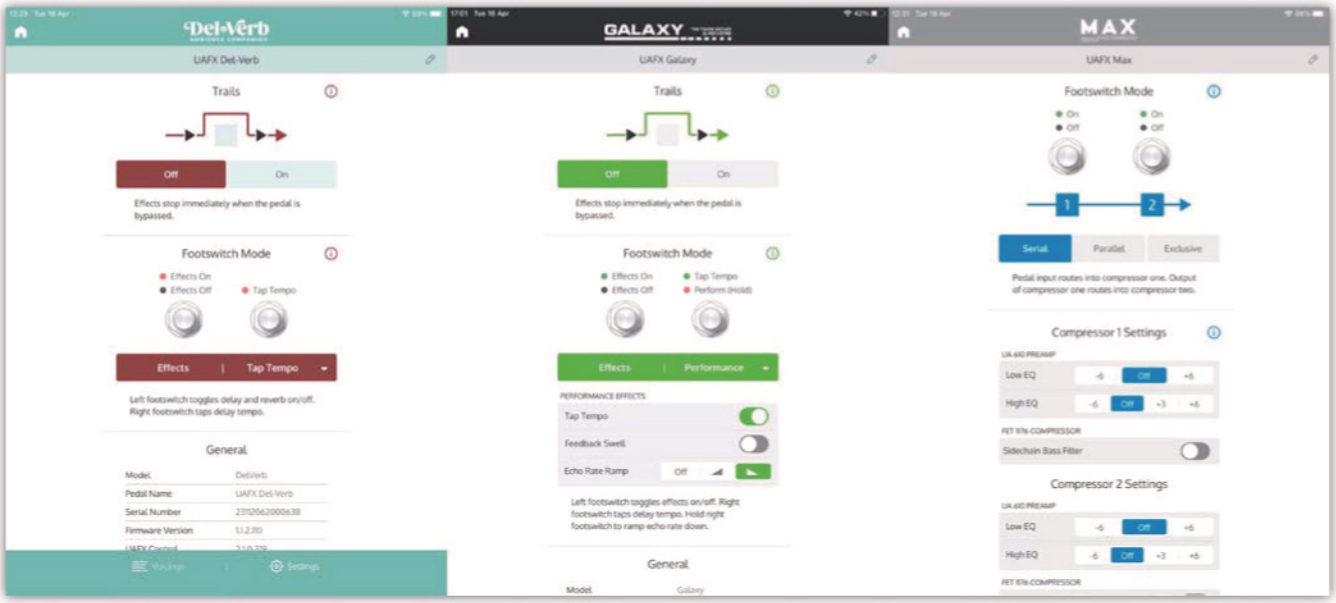

The UAFX Control app provides deeper configuration options and alternate assignments for the footswitches.

» inclusion of one here adds a bit more flexibility. Setting it in the middle matches the original hardware, and you turn up for a faster release, not a longer release time. Setting Ratio to off keeps the characteristic circuit colorations without applying any compression. The fixed ratio of a Dyna Comp pedal's response is maintained faithfully here, meaning that the Ratio, Attack and Release controls have no effect, triggering the 'UAFX rapid LED blink' reminder should you forget. The footswitch status LEDs double as rudimentary gain-reduction meters, changing from green to yellow to indicate a moderate amount of compression, and red to indicate heavier action. Not a precision tool, but certainly better than nothing.

Del-Verb Ambience Companion

No great surprise that the Del-Verb Ambience Companion offers "emulations of classic reverb and delay effects". Hold on, isn't that what Golden and Starlight do? Indeed it is, but here you get a bit of both in one pedal, albeit with less in the way of deep tweakery. Still, I'd say there is more than enough variety and flexibility in the Del-Verb for anyone who doesn't already have a Starlight and a Golden to seriously consider opting for this one instead, unless you really need to get in deep with the reverb settings.

There are two channels again, but this time one is dedicated to delay and the other to reverb. UA make no secret of the fact that these are the actual algorithms used in the Starlight and Golden pedals, so in the delay channel you have a hardware-switchable choice of a '70s Maestro EP-III tape echo emulation, an Electro-Harmonix Memory Man bucket-brigade analogue delay, and UA's clean, modern-sounding Precision digital delay. The reverb channel lists Spring '65, a tube-driven spring reverb tank from a favourite '60s American guitar amp, UA's now familiar EMT Plate 140 emulation, and the classic, era-defining hall and room sounds of a Lexicon 224 digital reverb from the early '80s.

What each of these choices actually does, however, both for the reverb and the delay, is subject to a wide choice of 'voicings' available in the app. Whatever you choose is then retained by the pedal for that switch position but is also subject to the settings of the physical controls. Not that there is a lot you can do to the reverb with just a reverb mix control to play with, but the sheer variety of reverb voicings available meant that I didn't feel I was in any way lacking in the ability to get something quite specific whenever I wanted it. What I did miss was a way to tweak the decay time without going back to the app for a different voicing. Where's that Main/Alt switch that the Galaxy has? On the delay side, however, if you inadvertently leave, say, Mod and Feedback turned up while you are auditioning app voicings, you may not actually hear what they are supposed to sound like as useful starting points; you get the combination of the voicing and the knob positions. The essential element of the EP-III's renowned preamp is included in the modelling here but, unlike in the Starlight, you can't retain just the tonal shaping of preamp with the delay mix turned all the way down. The delay and reverb are also always in series, with no option to run them in parallel.

The Del-Verb's delay time range is different for each emulated processor, with Tape EP-III offering 80-700 ms, Analog DMM 110-1068 ms, and Precision 120-1500 ms, set using the delay time control or, if assigned to tap tempo, the right footswitch. By default, this switches the reverb on and off, but if reverb is an effect you use subtly and tend to leave on permanently, repurposing the right footswitch in the UAFX Control app, using the left to switch delay only (or delay and reverb together) on and off, makes better use of the available capability. Feedback sets the number of repeats, with the Tape EP-III and Analog DMM able to go into self-oscillation runaway at high feedback levels, whilst the Precision will just repeat indefinitely. You can also ensure delay trails and reverb decay continue when you bypass the pedal by setting Trails On in the app.

The Color control affects the delay in different ways according to the selected effect: for the EP-III it sets the virtual tape-recording level, in the Analog DMM it sets the input gain, whilst in the Precision delay it simply sets the tone of the delay signal. Similarly, the Mod control gives the Tape EP-III the sound of a pristine, new tape set to the left, or older, more worn-out tape to the right, whilst in the Analog

DMM it offers vibrato to the left, chorus to the right, and off at noon. In the Precision delay this becomes a flanger turning left, and a chorus to the right. The pedal's dedicated Mix control only operates on the delay, as the reverb control already offers a reverb mix, but fully clockwise on either will give you 100 percent wet for use with an aux send.

Galaxy Quest?

Yes, it's a Space Echo emulation. Roland's RE-201 multi-head tape delay unit, incorporating a spring reverb, carved out a loyal user base from the mid-'70s that included not just guitar players but studio practitioners as well, especially those who appreciated its hands-on, real-time control of creative feedback and dub-genre spin echo effects. Guitar amps didn't have effects loops back then, so a Space Echo was originally always used into the input of guitar amps, which meant that the Space Echo's preamp also became a significant factor in the resulting sound. That's faithfully modelled here, and whilst you certainly can run this in an effects loop or on an aux send, I urge you to try it 'old school' into the front of an amp, real or modelled. What you lose in consistency, you more than gain back in expressive control, I find, and I don't think that's just because I, like many others, used delays that way for years when it was the only way you could!

The iconic big-knob head selector of the original Space Echo is replaced here by a mini-switch and LEDs, but all the combinations are there, with delay times that go beyond the original hardware: head 1, a dotted 16th-note with a delay range of 69-354 ms; head 2, a dotted eighth-note offering 131-672 ms; and the quarter-note delay of head 3, covering 189-974 ms. The 'magic' is in the combinations, as any pair or even all three can be selected together, with or without the reverb as well. The deep modelling includes all the variability of the mechanical aspects of the design, and the inevitable degradation of the tape itself that, together, resulted in a very musical modulation and tonal filtering. It's all wow and flutter, of course, but until it starts to get extreme, it just adds to the characterful sound that made these units so popular. The reverb on its own, being necessarily a short-tank unit just to fit in the chassis, is actually pretty coloured and trashy-sounding, but it plays an important role in the composite sound, making the repeats a little more diffuse, and I'm glad that its sound has not been 'improved', with springs and circuit having been accurately modelled from an original RE-201 specifically for the Galaxy pedal.

As it is a physical tape transport mechanism that's modelled here, changing the delay time with the Echo Rate control produces a change in pitch as the delays ramp up or down to assume a new tempo. The ramp times are, apparently, accurate to the original hardware, so they can take quite a while to settle. This can actually get quite annoying if you are just trying to dial in some conventional guitar delay effect, but is a major factor in the creation of soundscape-style special effects. Switchable might have been nice, though. Feedback sets the number of repeats, with the characteristic Space Echo self-oscillation on tap at higher feedback levels.

The remaining controls have dual functions, determined by the position of the Main/Alt switch. Input Vol sets the record level of the virtual tape loop, but in Alt mode allows you to choose the age of the tape, with higher settings giving the sound of more oxide shedding, wow and flutter. Dwell sets the drive level into the reverb tank, with the Reverb Vol control doing exactly what you expect on the output side. In Alt mode, however, Dwell and Reverb Level become Bass and Treble controls, affecting only the delay signal. The immediacy and effectiveness of the EQ was always a major factor in dialling in different sounds from a Space Echo, cranking the treble for a bright slapback, or rolling off both bass and treble for a long, rolling echo wash that doesn't get in the way. It is great to have it easily accessible here, with Alt settings, of course, retained when you switch back to Main mode. The right footswitch, too, has an extra trick up its sleeve in addition to its basic tap-tempo function: press and hold and you can temporarily push the delay time up or down or increase the feedback amount, or indeed both at the same time, according to what you have pre-selected in the app.

Is this the most accurate ever replica of a Roland Space Echo? I'm not sure I can say: I never had two real ones that sounded exactly the same anyway. We are spoilt for choice in this area, and for myself, I'd take any of the better pedal versions over the real thing these days.

Despite the limited controls on the Del-Verb pedal, there's plenty of scope to tailor the sound, courtesy of an impressive array of 'voicings'.

Where Next?

I guess you could say that this set of releases confirms UA's commitment to the UAFX pedal format. The continued omission of MIDI and any other form of remote switching says that these are, even more than the previous batches, all about simple setup, 'tweak to taste and use' functionality, like the majority of analogue pedals we have all used for years. It is just the fact that they are digital, and people know what the digital domain can do in terms of deep-level editing and real-time remote control, that perhaps creates a different burden of expectation. Viewed simply as meticulously accurate emulations of classic bits of gear that many people wouldn't otherwise have access to, does it really matter what else they do or don't do? Other products from other manufacturers offer the kind of remote control and preset structure that is clearly a benefit in certain live-performance applications, and I'm certainly glad they exist as well. But, ultimately, the sound is surely what we do this for? The sound is where the inspiration lies, and sometimes that's what matters most.

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RME Digiface AES

USB Audio Interface

Originally developed for a German broadcaster, the Digiface is now available to everyone.

HUGH ROBJOHNS

RME's Digiface range of USB audio interfaces probably needs no introduction to regular readers of *Sound On Sound*, but in essence these units currently all provide straightforward digital format conversion to/from USB. For example, the company's Digiface Dante, Ravenna and AVB models each provide 64 channels of audio-over-IP connectivity in different flavours — and both the Ravenna and Dante units also include another 64 MADI channels of I/O. For ADAT interfacing, the Digiface USB offers a quartet of Toslink connectors for 32 channels of I/O (am I alone in thinking Digiface ADAT would be a more descriptive name?).

The newest addition to this Digiface family is the Digiface AES — although it does rather more than you might expect from its name. When asked to review this new model I was expecting an uncomplicated 16- or 32-channel AES3 to USB interface as a logical extension to the Digiface family. However, it turns out that the Digiface AES is really a fully-featured 14-input, 16-output USB audio interface, and something much more closely resembling a Fireface product, really. In fact, comparing its capabilities and functionality I'd say the Digiface AES sits comfortably between the Babyface Pro FS and the UCX II.

As a quick overview, then, the Digiface AES features two mic/line analogue inputs and two balanced analogue line outputs, plus a stereo headphone output. On the digital side it possesses coaxial S/PDIF in and out, ADAT in and out (reconfigurable as optical S/PDIF), MIDI in/out and... oh yes, a stereo AES3 I/O as well.

Naturally, the unit enjoys RME's excellent SteadyClock FS technology, and also includes two channels of sample-rate conversion assignable to a selected stereo digital input, allowing a non-synchronous digital source to be mixed with other clocked digital sources. It is also fully class compliant, can work in a standalone mode, is bus-powerable, can be controlled from RME's ARC, and comes with TotalMix FX and DIGICheck applications. The RME driver allows up to three supported interfaces to be used together (provided they are clock synchronous).

If the extensive overlap of facilities with existing products seems odd, it's because this new Digiface AES was designed and built specifically as a custom product for a large German broadcaster whose requirements included easy connection to AES equipment, with bus-powering, mic inputs and high-quality monitoring. It wasn't originally intended to be a 'mass-market' product at all, but someone at RME saw the potential in it as a consumer interface and so it is now available, albeit at a price which reflects its slightly specialist origins (and possibly smaller-scale production run).

Controls & Connectors

Looking at the hardware in more detail, the Digiface AES is built to RME's familiar

format, in a 1U high, half-rack-width chassis. The left side of the front panel features a pair of combi XLR mic/line inputs with the control section on the right comprising a compact colour display, value dial and four direct menu access buttons. There are no high-impedance instrument inputs, but a single quarter-inch headphone socket is located within the blue control section just right of centre. Both the main XLR and headphone outputs are DC-coupled and can be used for scientific purposes or to create CV signals for synth modules. From an audio perspective, DC-coupling also removes the LF phase shift introduced by coupling capacitors and some argue this is audibly beneficial.

While the Digiface AES was designed to be bus-powered, it also features a bayonet-locking power socket for standalone use or where a host computer struggles to supply enough juice. A double-insulated, universal voltage, line-lump power supply is included, accepting a figure-8 (IEC C7) mains lead and providing a 12V/2A DC output.

Other rear-panel facilities include a Kensington lock slot, a USB 2.0 B-type socket for connection to a host computer, and a mini-DIN connector providing MIDI in and out (a breakout cable terminating in a pair of full-size 5-pin DIN sockets is included). There is no word-clock in/out provision.

The digital audio I/O encompasses a pair of RCA phono sockets for coaxial S/PDIF in/out, with a pair of Toslink optical ports directly above configurable either for ADAT or optical S/PDIF connections. Two XLRs provide the AES3 I/O, and two more deliver the analogue line outputs.

All standard sample rates are supported between 44.1 and 192 kHz over the AES3 and both flavours of S/PDIF, while the ADAT interface can use S/MUX protocols to transfer four channels at double rates, and two at quad sample rates. AC3/DTS-encoded surround material carried on an S/PDIF connection can be passed straight through to a home theatre decoder, if desired, although the unit's sample rate must be set to 48kHz.

In terms of technology the digitally controlled mic/line preamps are the same transformerless design developed for the Babyface Pro FS, with AutoSet automatic gain adjustment, individually selectable 48V phantom power, a manual gain range of 75dB (in 1dB increments), a relay-switched -10dB pad, and an EIN figure of -129dBu (A-weighted, 150Ω, 20Hz-20kHz). The mic input impedance is 2kΩ (rising to 5.3kΩ with the pad engaged), and the maximum mic input level is +18dBu (with the pad).

Line inputs present a 12kΩ impedance, enjoy the same gain range, and accept a maximum input level of +24dBu. When selected, phantom power is only present on the XLR terminals, not the inner TRS socket. Usefully, every physical input and output also features a fully-featured three-band parametric DSP EQ section.

The electronically balanced main outputs are switchable to any of RME's three standard operating levels, aligning peak digital level (0dBFS) to either +19, +13, or +4 dBu (all from a 300Ω source). I was surprised that RME haven't included a +24dBu output level option (which is now offered on other recent RME interfaces). However, the +19dBu option obviously meets the needs of German broadcasters so I presume a +24dBu mode wasn't considered necessary in this custom product. There's certainly more than enough output level to drive active monitor speakers, though, which is what most users would probably need them to do.

A point worth noting from the generously comprehensive 104-page »

RME Digiface AES
$1999

PROS
- Bus-powered.
- Sample-rate converter allows mixing of non-synchronous digital sources.
- Remarkably low analogue latencies.
- Excellent D-A performance.
- Three-band parametric DSP EQ on every channel.
- Standalone operation.
- TotalMix FX.
- DC-coupled outputs.

CONS
- No high-impedance instrument inputs.
- Price.

SUMMARY
A well-featured compact interface with a unique selection of facilities to set it apart from other RME interfaces.

The Digiface AES's back panel features a USB-B port, a socket for the breakout MIDI I/O, ADAT and S/PDIF I/O, and AES and analogue audio I/O on XLR sockets. A pair of combi XLR inputs and a headphone socket are found on the front panel.

» manual is that when driving an unbalanced destination the XLR pin 3 (cold) terminal should be left floating. Shorting pin 3 to ground — as most ready made unbalanced adaptors do — apparently results in increased distortion.

The headphone output is switchable between high and low power modes equating to a maximum signal voltage of either +13 or +7 dBu, delivering 160mW per channel into 32Ω phones (with 0.1% THD and from a 5Ω output impedance). For comparison, the UCX II provides a beefier 210mW per channel, but I didn't have any practical issues in the studio with headphone volume using AKG K702s or Sennheiser HD600s.

As usual, I ran my standard AES17 dynamic range test on the Digiface converters, as I find this to be a fair indicator of overall technical performance. My measured results are very similar to the published specs, with a figure of 119.8dB (A-weighted) for the D-A converter, which is an excellent result, and similar to other recent RME products.

For the A-D stage the measured figure was 114dB (A-weighted), which is similar to the Fireface UCX II but 5-6 dB lower than more recent RME products (for example, the AIO-Pro measured 119dB and the 12Mic 120.4dB. In terms of relative performance, the Digiface AES's A-D dynamic range lies roughly midway between that of the Audient ASP880 and the Ferrofish Pulse 16 — so certainly not bad by any stretch. RME tell me that the limiting factor in this dynamic range measurement is the mic preamp itself — the A-D converter could manage 8dB more range — but apparently ultra-wide dynamic range through the analogue inputs was not a customer requirement when designing the Digiface AES.

Talking of converters, the devices employed in the Digiface AES are ESS Sabre types (ES9842Pro A-D and ES9080Q D-A), both configured with minimum-phase filter topology (like many other RME products). This delivers a much shorter latency than traditional linear-phase designs, with only 0.07ms (3 samples) for the A-D and 0.38ms (17 samples) for the D-A, at a 44.1kHz sample rate. For comparison, most conventional converters (using linear-phase filtering) introduce more like 0.75ms of filter latency at base sample rates.

"The very low A-D converter latency and bus-powering will undoubtedly be a big plus for many users."

Operating the Digiface AES is much the same as the Fireface models, with most things being configured through the front panel, while others are set using the TotalMix FX application on the host computer. Different I/O configurations and setups can be saved to six preset memories, allowing instant recall for different recording or playback requirements, and these can also be accessed in standalone mode. In other words, handy setups can be pre-programmed for use away from the computer, and the (hardware-based) TotalMix functions can be controlled remotely via MIDI, if required, which is a nice feature in standalone mode. As you would expect, the front-panel data knob not only accesses key configuration parameters, but also serves as a volume control for both the main analogue balanced outputs and the headphone output, switching between them by pressing the knob.

Conclusions

Given the immense capabilities of the Digiface AES, I remain surprised the product isn't marketed within the Fireface interface family. After all, it looks more like a Fireface than any of the current Digiface models, it has analogue I/O, and its facilities overlap the existing UCX II to a great extent, even sharing the same complement of digital I/O.

That said, the Digiface AES has distinct features that set it apart from the UCX II, such as its two-channel sample-rate converter allowing two non-synchronous digital sources to be mixed together, and its superior D-A performance. The very low A-D converter latency and bus-powering will undoubtedly be a big plus for many users, too.

The UCX II's strengths lie in having eight analogue inputs and outputs, with high-impedance instrument facilities. It also has word-clock in/out to help with external equipment synchronisation, a more powerful headphone amplifier, internal DSP allowing reverb and echo in TotalMix FX, DURec recording direct to a USB drive, and it also costs significantly less than the Digiface AES too.

So why would you choose the Digiface AES over the UCX II (or Babyface Pro FS)? That's a tough question, just because they overlap in capability so much, but a choice would come down purely to specific application priorities and requirements. We're very lucky to be offered such alternatives!

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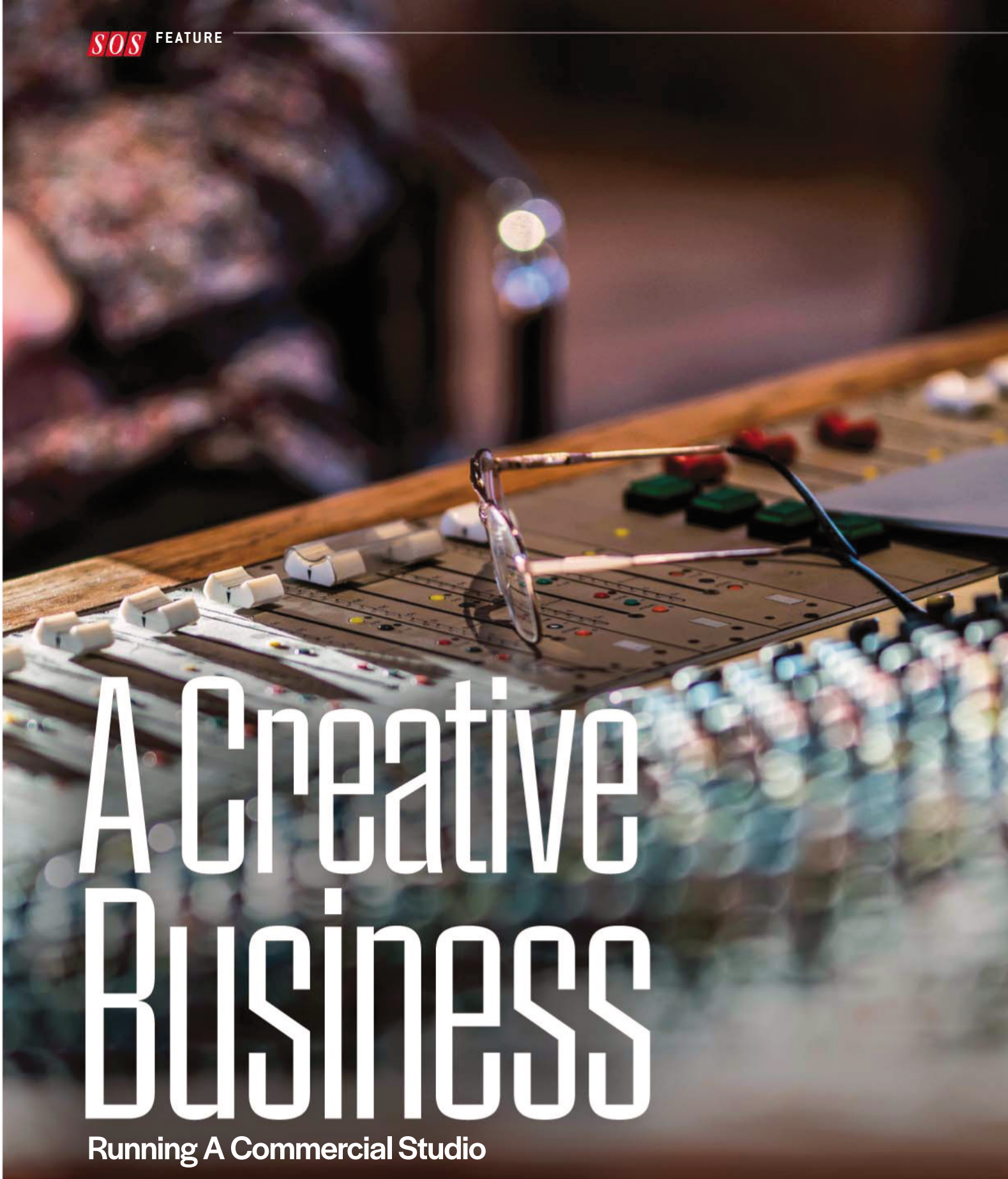


A Creative Business

Running A Commercial Studio

Many of us dream of setting up a recording studio. But where do you start — and what's it really like to run one as a commercial business today?

Photo: WeTheDee/StudiOwz

NEIL ROGERS

We've often shone our spotlight on the world's top recording studios, treating you to gorgeous pictures of great spaces and high-end gear, and recounting tales of sessions with big-name artists. In every country, though, there are also numerous smaller commercial studios that may not achieve Abbey Road levels of fame or income but succeed in making their owner-operators a living. I run such a studio in Cambridge. Recently, I've been thinking a lot about how I can develop the business so that it remains relevant in the years to come, and decided I'd pick the brains and experiences of some other studio owners in different parts of the UK. They've developed different approaches to their businesses over the years, and I found plenty in our conversations to mull over — and if you have dreams of running a studio as a business, I hope you will too. You can find out more about each studio in the boxes that appear in this article, so let's dive in and explore the reality of running a studio as a creative business in 2023.

Dreams & Reality

Nobody in their right mind would go into the recording studio business today to make their millions! So I was curious to learn how and why these studio owners ended up doing what they do, and what motivates them to keep going. For Pete Fletcher of Black Bay Studios, the motivation is clear: "I love making records. It's thrilling, rewarding and challenging. Making an album in a big studio in a crazy location is many musicians' idea of 'living their best life', so it's incredible getting to share those experiences with people... though there's some pressure in that too; you want to get everything right for people!"

But why set up a studio instead of working freelance elsewhere? "When I started to take the idea of being an engineer more seriously there were limited opportunities for me to work in commercial studios, so I pursued a DIY approach: basically, rehearsal rooms with some basic equipment in the corner. What I have now »

Boe Weaver, aka Jim (left) and Rob Holmes.

Photo: WeTheDee/StudiOwz

Jim Holmes: "You can have a world-class recording chain with a single swipe of your credit card — you have to be able to offer something more than just nice equipment."

Advice For New Studios

Could our studio owners offer some tips for anyone wanting to turn their studio startup dream into reality? Owain posed a question: "Do you love it? I have a classic car and it's my pride and joy. Someone once told me, referring to the car, that 'you'd better love it, because if you don't, with the amount of problems it's going to cause you, you'll end up hating it!'" Jim had similar thoughts: "The pressure and stress of any creative business can be a killer, and I think you have to be the right kind of person to run a studio... If you don't like dealing with people — difficult ones specifically — then I'd avoid it entirely. You also have to love music, because otherwise it can be hard to justify the all-encompassing demands on your time, and time away from family and loved ones." Dan advised that "it takes a long time to build up a client base. You work on some awesome music and you want to tell everyone about it, but the label doesn't release it for another nine months! In the beginning, be prepared to have a side hustle, just to keep yourself financially afloat." For those who remain undeterred by all these words of warning, Pete said to ask "what's the building like? I'd be thinking about acoustics, basically: ceiling heights, layout etc. You can always upgrade a microphone but it's hard to make a space sound amazing if it's not a good starting point. The other question is 'do you *need* your own studio?' Hiring someone else's space or partnering with an existing space can be a great option too."

» has grown from there over many years, through various spaces in Nottingham and then, eight years ago, to the Outer Hebrides to follow a 'residential studio by the sea' dream."

Owain Fleetwood Jenkins, owner of StudiOwz in Wales, also forged his own path: "After four years studying music and technology in Birmingham, I applied to 30-plus studios for work as an assistant engineer. I had zero responses, so decided to pave my own way. I began recording local bands with the limited equipment I'd started to collect. After a few years, word spread and I was working on bigger and bigger records." Now, he said, "we've built a studio that inspires me to make the records I want to make, in the way I want to make them. I truly enjoy working and living with bands in our residential space. Constantly meeting new, wonderful and talented people is a joy to call 'work'."

The Holmes brothers at Empire Sound on the Isle of Wight started their original studio off the back of an old band project, recording themselves before going on to record other people. Jim was pretty blunt about why they do it: "It's not the money! I think there can only be one good answer to that question... we genuinely love recording music. I think you have to!" He added that "we have always enjoyed getting a better result than an act was expecting. If you can also help them have a great time while they do it then that's our goal; the music industry will add its own stresses and pressures soon enough and, as much as possible, we try to keep our clients away from that world whilst recording music with us."

Dan Cox of Urchin Studios in London also loves making records, but the variety appeals too: "Running a recording studio means every day is different! One day you're mixing, the next you're engineering a band, the next you're setting up with another producer for a vocal session. This is a people business, and hearing each and every artist's story is part of what makes it so worthwhile." Yet, all of the owners were clear that their studio must be more than a creative indulgence, and Dan put it succinctly: "Anyone who runs their own business knows that it's a huge commitment and a lot of responsibility. The fact we're in a 'cool' industry doesn't take away from that. There's tax, fire regs, company accounts, refuse collection, chasing invoices and all of that mundane stuff that's a drain on time."

Jim was also clear that: "100 percent, we have to think of running the studio as a business. At this stage in our career, we have no management helping us, so every single act that comes to Empire has been dealt with, chased and seduced by us!" For Pete, it's very much a case of balancing the creative and business sides: "I was already several years into running my first studio as a limited company when someone referred to me as a business owner. It was a huge shock, and definitely not something I saw myself as. Now it's much more like that, [*but*] I'm still, without doubt, running a business to be able to pursue making records as a career, not the other way around."

Pete Fletcher, owner of Black Bay Studios.

Photo: Neelam Khan Vela

Pete Fletcher: "Our USP is the setting, so it was important for us to have large windows and generally make the studio feel part of where it is."

Owain pointed out, though, that you can still allow room for individuals to focus on the creative side: "Mostly, I still see myself as a 15-year-old school kid on work experience, but hosting and running studio sessions has always felt natural to me. As far as invoicing, doing my tax return, and organising the calendar — the things I call 'adult jobs' — not so much. It's not a one-person show here. There's a team of us ensuring things run smoothly, from booking bands and catering, to running the residential side."

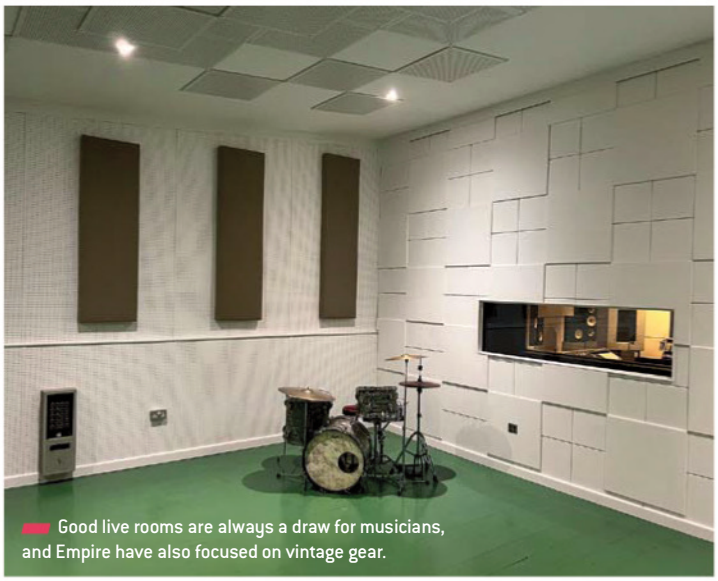
Not My First Rodeo

These guys all had previous experience before setting up their current businesses, and I asked how this had informed development of their new studios. "My previous studio", said Owain, "really helped me concrete the idea of what my 'dream studio' would look like, and how it would function. I wanted to make sure I put things in place to make my day-to-day life a little easier — even down to the simplest things, like having plug sockets at your feet in the control room." He explained that he'd cut his teeth running a studio in an old cow shed on his parent's farm, where he developed his engineering skills and learned some of the

Empire Sound

Brothers Jim and Rob Holmes own and are full-time producers at Empire Sound, a recording studio on the Isle of Wight. They've run a studio together for 19 years, and I visited their previous one, which was situated in a Victorian water tower. In a sadly familiar story for creative businesses, a large rent hike in 2020 led them to relocate — but the silver lining was that the move gave them the chance to re-evaluate how they saw their studio functioning. Empire Sound is now located in an industrial estate, and Jim explained that they like being inconspicuous from a security perspective, but that they also enjoy seeing the effect on clients when they walk through the studio's doors into the "ridiculously stylised" environment, which has a minimal, vintage aesthetic. The studio also reflects how the duo like to produce music, with an emphasis on capturing musicians playing together, committing to sounds 'on the way in' and generally being bold when going for a particular vibe. The new studio's live room is key to this approach. "The space we took on had originally been a studio of sorts," said Jim, "and our original plan was to completely change the ceiling height and make other structural changes. We thought we should at least try out the room first, however, and immediately fell in love with how the room sounded... The goal then became subtly refining the room without ruining it!"

W https://www.empiresound.co.uk



Good live rooms are always a draw for musicians, and Empire have also focused on vintage gear.

realities of running a smaller-scale studio. Whilst StudiOwz is on a much grander scale, Owain approached development of his 'dream studio' with a clear idea of what his clients would want: "We're based in a beautiful rural location that will appeal to people wanting to get away from the city. It was essential to me, though, that the studio had that 'wow' factor that also delivered the goods once people got to work".

Jim thinks of his previous studio as "a 19-year-long case study in what not to do! Don't get me wrong," he explained, "we did get many things right, but addressing our existing problems was not only cathartic but [*also*] a brilliant foundation for developing our new space. Some problems, line of sight for example, are a simple fix. It was more issues with the flow, layout and busyness of the space. We were keen for Empire Sound to be as streamlined as possible." Another conscious choice was not having sofas or furniture that turned the control room into some kind of lounge/chill-out space: "We stop short of telling people to stop looking at their phones, but we want to do what we can to make the control room a space where everyone is engaged in what is going on."

Dan was also keen to address historical problems: "One thing we struggled with in the old studio was noise transmission between the two studios. In building a new complex of 11 rooms, we had to focus on getting that right. This meant getting one of the rooms to the point of being able to get a door on it, sticking a drum kit in there and then bashing the bejesus out of it to see what was spilling out into the next room! In the case of this test, it wasn't initially quite up to scratch, so we ended up putting in additional walls to make the soundproofing as bomb-proof, or drum-proof, as possible."

Pete explained that he'd wanted to retain the 'heart' of his old setup: "What was great about [*my previous studio*] First Love was that it was full of weird and wonderful instruments, and [*had*] a big live room and... an environment that bands felt comfortable in. That's definitely something I've tried to bring through. Black Bay is the product of about 20 years of daydreaming about the perfect setup."

With any small business that runs out of a brick-and-mortar facility, it's essential that the rent or mortgage, which is the biggest overhead, is not only affordable but also secure. Rented buildings can be the only viable option in expensive areas, in which case you'll need some assurance that your landlord won't suddenly put up your rent — this is something Jim experienced at his previous studio. "With our new space," he told me, "it was essential for us to have some long-term security. We took our time finding our current location and, whilst still renting, we found an affordable situation that we could feel confident investing in for years to come". The team at Urchin had a similar experience, and Dan observed that "when we embarked on our new studio, we could only do so with the security of a long-term lease that we felt was affordable over a number of years.

»

StudiOwz' tall former chapel building allowed Owain to create a stunning mezzanine control room.

Photo: WeTheDee/StudiOwz

StudiOwz

Run by engineer Owain Fleetwood Jenkins and his partner Jodie Marie, StudiOwz is a residential recording studio in Pembrokeshire, Wales. The main building is a converted Baptist chapel built in 1804, and it has a cottage next door that can accommodate bands and musicians. The conversion from derelict chapel to studio is a striking one, and as someone who's done plenty of dry-walling and building work in my own studio, I was hugely impressed to learn that Owain had, with the help of friends and family, dry-lined the entire inside of this large space. The studio adheres to the original shape of the building, but now includes a couple of isolated areas and a mezzanine-style control room that looks out over the recording space. It's a well-thought-out space for recording musicians, who seem to appreciate Owain's decision to focus his early investment on acquiring an eclectic selection of backline and microphones, rather than filling endless racks with outboard. Owain's vision was to create "an inspiring place" for bands, where they could "get away from everyday life and focus on recording their music". Typically this sees bands and artists using StudiOwz for tracking and then taking away their projects to finish and mix in their own studios. StudiOwz has now been open for four years, and these early decisions seem to be paying off: the studio is attracting a steady stream of bands and musicians from the larger English cities such as Birmingham, London, Liverpool and Manchester, who are making a choice to travel and 'get away' from their own environment to focus on recording a project.

W https://studiowz.co.uk

» We've put substantial investment into the new studio and it would have been crazy to do so without taking due care."

Black Bay and StudiOwz are both in fairly remote locations, where suitable spaces are more affordable, and both Pete and Owain explained that they'd been able to buy their properties for relatively modest sums — but that even with a DIY approach it's hard to underestimate the costs involved in building and fitting out a professional studio facility once you have the building, even before you consider the gear.

Biggest Challenge?

I asked the owners what they feel are the biggest challenges in running a studio today. For Dan, "it would have to be the day-to-day plate-spinning. Our current setup is a far cry from our single-room studio back in Limehouse in 2007. Some days there's just a bit much on your plate... you've got a full recording session to do but also one of our tenants reports that the front door won't open or there's no toilet paper!"

Dan Cox (left) and Matt Ingram, of Urchin Studios, London

Dan Cox: "Getting out to gigs, music industry events and inviting people over to the studio is really important."

Jim suggested that "location is our biggest issue. Although the Isle of Wight is a wonderful place to take a moment away from the real world and make a record, we don't operate a residential studio and not everyone has the time to 'get away'."

For those who do run residential facilities, the hardest aspects of studio life are probably not a surprise. Owain explained that they "have bands come and stay for weeks at a time. It's hugely satisfying seeing all our hard work being used to make music, but it's also a lot of hard work having people stay in our home — it really is an all-consuming lifestyle!" Pete focused on some of the less obvious costs: "Our electricity bill is horrendous, particularly as our heating is electric. We have a really big space, which is amazing, but it all needs to be warm. Cleaning is a huge undertaking too. We have nine beds, three bathrooms and a kitchen... It's about eight hours' work to get the place turned around for the next clients, and I often feel like I'm running a B&B and not a studio!"

Studios of this size generally can't afford their own maintenance engineer, and every one of the owners identified maintenance as a significant issue. Pete »

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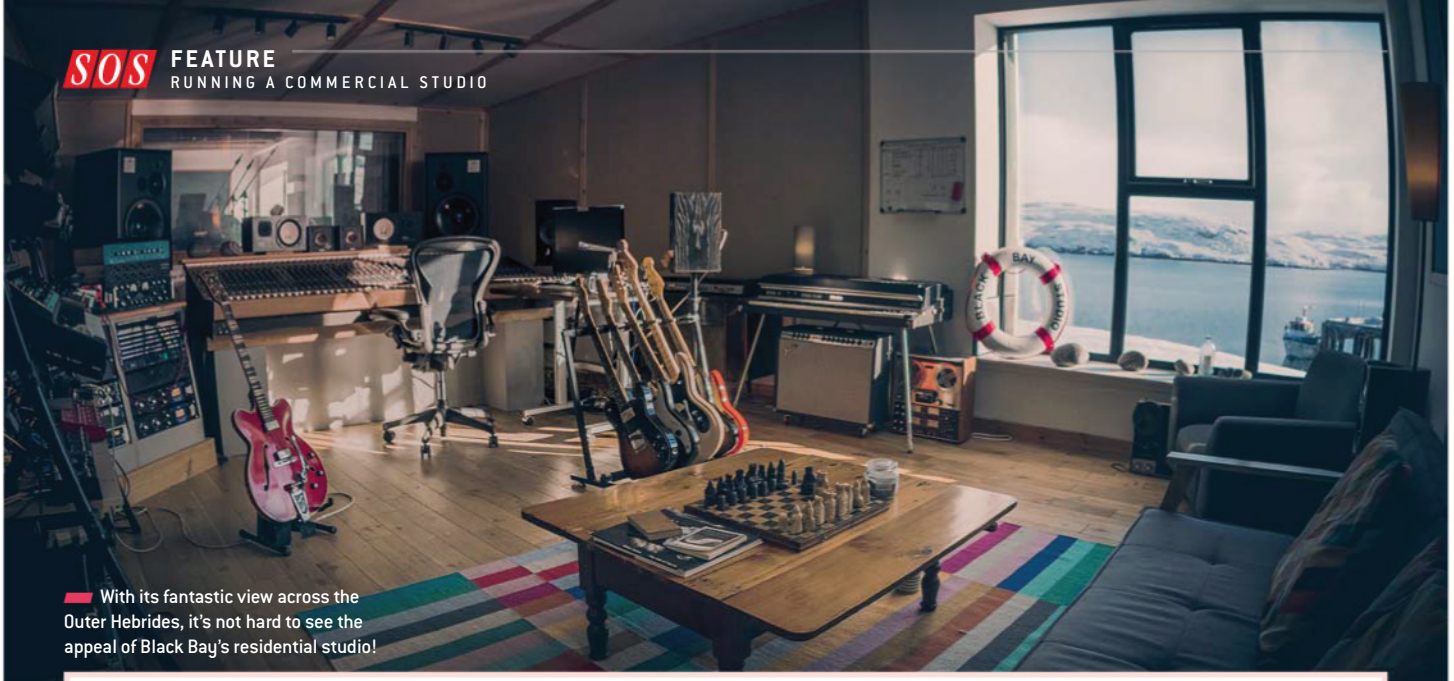

With its fantastic view across the Outer Hebrides, it's not hard to see the appeal of Black Bay's residential studio!

Photo: Neelam Khan Vela

Black Bay

Situated in a former crab-processing factory in Scotland's Outer Hebrides islands, Black Bay Studios is one of the most remote recording studios in the UK. Indeed, owner Pete Fletcher describes it as a "residential studio at the end of the world". It's an impressive setting, and Black Bay seems to tick all the boxes for an 'escape' studio, offering comfortable, good-quality accommodation, as well as a great studio.

The story of Black Bay's inception and development offers a glimpse into the sort of issues you can encounter when chasing a studio dream like this. Pete explained that he ran a small studio in Nottingham for several years before his "partner got offered a job opportunity in Stornoway, the capital of the island of Lewis and Harris, and my thinking was that there's bound to be some cool old building waiting to be converted into a studio there!" It took real ambition and dedication to get this project off the ground, though. On moving to the island in 2015, Pete drew on his existing client base to continue mixing and mastering in a small home studio, while he searched for the right building. The reality was far harder than he'd anticipated. "When it comes to investing all of your money into a project," he told me, "you have to be able to step back and distinguish between 'the dream' and 'reality'. It's hard to underestimate just how remote we are here, and I was clear in my mind that this idea would only work if the studio was on a scale impressive enough to encourage people to come and record here." Sadly Pete "didn't initially have enough money to do that" but, rather than give up or blindly plough ahead, he researched possible sources of funding and, with hard work and perseverance, secured financial help from Scottish and EU schemes offering support to businesses in challenging locations. Setting up not just a studio but also the necessary accommodation plunged Pete into a world of permits and planning permission: it took him over a year to find the building and secure the necessary consents, and then another of building work before the studio could open in 2017.

The hard work and business-minded decisions seem to be paying off, as the studio has gained a reputation to rival some of the most famous residential studios in the UK. They host a lot of bands from the big Scottish cities, but also from the wider UK and further afield, with clients from Europe and beyond. And well-known producers, impressed by what Black Bay can offer, are now bringing acts in. It was also great to hear how the studio has been supported and welcomed by the local music community.

W https://blackbaystudio.com

» confided that "it's actually a nightmare. We do a lot in-house, but essentially everything is constantly in the process of being about to break at any moment. That's true of old stuff, new stuff, expensive stuff and cheap stuff, and you do have to budget for it, particularly when you're on a remote island in the Atlantic! I have three working pairs of massive power supplies for the desk, for example." StudiOwz has lots of lovely classic gear from the '70s and early '80s and Owain told me that it "does have its drawbacks. At 40-50 years old, it seems the time has come when a lot of that equipment needs upgraded parts or recapping, on top of regular wear and tear. I've always found it frustrating when renting other studios and the Hammond organ doesn't work, or the pianos are out of tune... I promised myself I'd always keep on top of these things."

To keep a lid on the costs, Jim at Empire said they'd "learned to do some of the maintenance ourselves, but with anything valuable or even slightly tricky we will send it to the best tech we can afford. We are obsessed with having everything working to its best level and, truthfully, we spend a good whack on maintenance every year." The guys at Urchin also "do as much as we can in-house, partly because it's cheaper than hiring somebody else, but also because [*we*] find it enjoyable to learn new things, and developing these skills will hopefully help us keep the studios in good shape long-term. Of course, things break that you can't fix yourself... Off goes the AC30 to the amp doctor, again!"

Attracting Clients

I was especially keen to hear what the owners believe are the most effective ways of attracting clients, and it all starts with being clear about what the studio offers. As Jim put it: "We live in a time where you can have a world-class recording chain with a single swipe of your credit card — you have to be able to offer something more than just nice equipment. We're passionate believers in 'doubling down' on what you're good at genre-wise, and not trying to create a studio or produce music that is necessarily suitable for everyone's tastes. The best experiences we've had creatively and from a business point of view are when we have stuck to our guns and produced music how we like to make it."

For Dan, the location is a crucial factor. "Hackney Wick is a thriving creative community in London, which is obviously one of the world's major music cities. Putting our studio in the thick of this environment means we're surrounded by like-minded music-makers and a creative community." Location is important for Pete too, but for different reasons. "I think people come to us initially for an adventure... Our USP is the setting, so it was important for us to have large »

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» windows and generally make the studio feel part of where it is. Once here, they find a great studio and an environment that encourages creativity, and that's what brings people back or leads to recommendations... People primarily use us as a tracking studio, so it was also important for us to have varied and good-quality backline, such as guitar amps, pianos and interesting or fun keyboards for musicians to play and get inspired by." Owain made a similar point: "People are coming to a studio like ours to get away, get creative and record, not to mix or get into the super-fine details." He too touched on the gear: "Investing in backline was one of the best decisions we made from a business point of view, and we get clients booking our studio specifically to use our piano or Hammond."

There then comes the question of marketing, and Dan explained that "networking is so, so important, especially with the studio being based in London... Getting out to gigs, music industry events and inviting people over to the studio is really important. Even if it's music-makers directly, you never know who's got a friend that needs to make a quick demo, that results in them coming back six months later to record an album." Jim explained that networking is rather more challenging on the Isle of Wight: "We have a network of people that we work with but we have to work hard to stay on the radar for labels and independent artists and encourage them to come over to 'the island' to work with us."

Pete stressed that "making a great record is still the best marketing you could ever do... and a lot of people seem to have found us through the various projects that we have been involved in." Jim concurred: "Being on a smaller island community means that word of mouth is everything: If you do a good job *everyone* will know — but this also works the other way around! We've learnt the hard way that you should only take a job if you feel you can achieve their goal and it's important to treat everyone equally and never badmouth other clients. A good reputation is the golden goose." Owain added: "There's a saying that goes something along the lines of 'you're only as good as your last record'. I don't think that's strictly true, but it's really important to remember that, whatever you're working on, you need be excited, enthusiastic and working to the best of your abilities."

Owain Fleetwood Jenkins, owner of StudiOwz in Wales.

Owain Fleetwood Jenkins: "I've always found it frustrating when renting other studios and the Hammond organ doesn't work, or the pianos are out of tune... I promised myself I'd always keep on top of these things."

Owain also discussed a different tactic. "The majority of marketing for the studio is through social media, but we do advertise in *TIWN* magazine, which is a free Welsh magazine." I invited him and the others to share more about how they viewed and were using social media. "Using a really good photographer — ours is called WeTheDee — has helped when posting to my socials," said Owain. "I think it's important to maintain a presence to keep in the back of people's minds. You can't force people to come and record with you but, as long as they know you exist, when the time is right they might just reach out." While it's less central to Urchin's approach it's still important, as Dan explained: "When we started the studio it was before social media had even taken off, so all we had was word of mouth and that still feels like our most important marketing tool. The big difference now is that if somebody hears of us in conversation, they will almost certainly check out our Instagram before getting in contact, so we do try to maintain an active presence." Jim believes that: "having a good presence on all the social media channels is important, but the most positive aspect that we have found from using these tools is that it's a great way of reaching out, networking and generally looking for new opportunities."

Minimising Downtime

Few music studios will be fully booked all the time, and there will be times when you need to be elsewhere, doing other things. It makes sense to make that studio downtime productive if possible — there are plenty of potential avenues a studio would be well-placed to explore, such as podcast production and live-streamed gigs, and I was curious to know what, if anything, our studio owners were doing along these lines. What struck me most about all their responses was that the owners had chosen options that didn't spread their time and skills too thin: occasional events, or long-term sub-lets and contracts that mean they don't have to duplicate their efforts when it comes to things like marketing.

In a way, Urchin Studios were designed from the ground up to side-step this issue, with multiple spaces being let out to other artists and producers. But Dan also explained that Urchin co-owner Matt "has been running a drum recording workshop a few times a year, and I've been working with a mentoring scheme locally. They're not things that supplement studio income

Urchin Studios

Engineer/producer team Matt Ingram and Dan Cox have run Urchin Studios since 2007, and the current premises in Hackney Wick, London is its third location. They moved here after the landlord of their previous home of eight years decided to sell up to a property developer; an all-too-common story for creative businesses in London! Happily, the new building inspired ideas for a different way of running and sustaining their studio business. Today, Urchin is representative of many of London's studio spaces, in that it has a main, traditional recording studio (Studio A, with a good-sized live room featuring a drum isolation space, its own small toilet and chill-out space), but also several separate, smaller 'production' studios that are rented out full-time to professional musicians, engineers and producers. The interior is striking, with huge windows (which, fortunately, were already well soundproofed when they moved in!) flooding the building with daylight, and communal spaces helping to realise the pair's vision for this studio as a music 'hub'. With a recent nomination for MPG Recording Studio Of The Year under their belt, it seems the pair's perseverance and hard work has got the third incarnation of Urchin Studios off to a great start!

W www.urchinstudios.co.uk

At Urchin, a traditional main studio is joined by several smaller spaces, and benefits from plenty of natural light.

directly, but they do mean new faces through the door and often that creates new leads for work."

Owain, whose overheads in rural Wales are obviously lower, also values events that get people through the door: "When we first opened and had plenty of downtime, we would put on live concerts to an audience of around 35 people. Although this was a loss leader, it was very successful in securing new bookings for the studio and spreading the word". Building the studio business in that way was a success, and Owain finds he can now focus largely on more traditional recording studio projects.

Pete at Black Bay also seems to have reached a level where he can focus on traditional music recording sessions, but said that they "do some workshop-style songwriting camps, which are great fun and a good way of promoting the studio to musicians. I'm here to make records, however, and it's easy to get pulled away from that."

Jim at Empire Sound remarked that: "We have a pretty equal split between running the studio as a 'for hire' space and a place for us to work on soundtracks, scores and library albums, alongside our production work. We have a new relationship with the local music college that we're excited about, however: they are taking a few days each month to incorporate using the studio for their degree and masters students, and it's a welcome bit of fixed income for us."

The Future

Finally, given the rate of change in the recording world, how do these studio owners see recording studios adapting in the future? "There will definitely be technological [*advances*] in the next few years that may make some of what we do redundant," said Dan, "but musicians still need collaborators and will always value great spaces to get together and focus... We do need to be adaptable, though, and to keep an eye on the way music makers are working."

Owain felt "optimistic about the future of recording studios. We started out with analogue recording studios, then came digital, and now we're in a place where we've combined the two. I think this relationship is only getting better and more exciting. I also see first-hand the enjoyment people have from being here and I believe musicians will always search for that experience, time and time again. Coming to the studio is a special occasion."

Pete pointed out that "there's more to making a record than just the mechanics of putting some parts down, and I think a studio and producer can elevate it to a place that is beyond what you imagined. A good studio provides inspiration (as well as a proper drum sound!). Who wouldn't want to spend a couple of weeks making a record in a big studio?"

Jim also believes that "there will always be a market for recording facilities, but I think the ones that will survive will be the ones doing what they do very, very well. People aren't going to pay for something they can do with an interface and a laptop — why would they? I am confident there's still going to be a place for studios like ours though!"

PUSHING THE ENVELOPE

Using Transient Processors

Attack-enhancing plug-ins can transform your mix — but they need to be used wisely!

MIKE SENIOR

One of the more common reasons for processing a track at mixdown is because you want to reshape its attack in some way. Your kick drum might feel like it lacks punch, say, or you want more 'smack' from your snare. Alternatively, maybe your percussion feels too spiky and upfront, or your piano part feels overly hard-edged and aggressive.

Now, a lot of project-studio owners instinctively reach for EQ or compression to address these kinds of issues, but those tools aren't always well suited to the task. So you might try to boost your kick drum's low midrange to add punch, but discover that this makes the drum's tone horribly woolly into the bargain. Or you might cut some high end from the percussion to soften its attack, only to find that it now sounds dull and congested. Many producers have also noticed that slower-attack compression can enhance a snare drum's onset level spike (often referred to as its 'transient') by allowing it to zip through the processor before the gain reduction has a chance to react. However, because compressors work according to a fixed threshold level, this trick quickly becomes rather inconsistent with drum parts that have any dynamic variation to them — typically, louder hits will get more attack enhancement than quieter hits, which may not be what you want! Likewise, fast-attack compression is often used to reduce attack, but if you try this kind of treatment on a hard-edged piano part, you'll rarely be able to bring the compression threshold down far enough to soften the transients of lower-level notes without unconscionably level-pumping or distorting the rest of the instrument's sound.

Fortunately, there are specialist processors that can avoid many of these problems by approaching transients in a different way: not in terms of frequency response or signal level, but more appropriately in terms of how fast they cause a signal's level to change. This kind of processing incurs far less 'collateral damage' on the rest of the signal, because you can cut/boost just the transients, without affecting the sound's overall timbre or dynamics. And because a signal's rate of level change is independent of its absolute level, transients are detected and processed the same irrespective of whether they're loud or quiet, which means you get more consistent attack shaping

Threshold-dependent Processing

In this article I'm discussing threshold-independent transient processors, because these are the most common ones you'll find these days. There are some transient processors, however, that actually do work according to a fixed level-threshold — so whenever a drum signal exceeds the level threshold, a tiny gain spike is triggered to boost/cut what is usually the onset of the drum hit. Although this approach can create very similar attack-adjustment results for unvarying drum samples, any drum part with musical dynamics can produce a rather hit-and-miss outcome here, so I'm not a huge fan of threshold-dependent transient processing for general mixing purposes. I certainly wouldn't spend money on such a plug-in, as it's very easy to recreate very similar functionality by setting up a fast-acting gate as a send effect: whenever a drum hit exceeds the gating threshold, it lets though a short burst of drum transient which you can simply mix in to bolster the attack of the unprocessed signal to taste.

The granddaddy of transient processors, the SPL Transient Designer hardware outboard unit, in its current MkII incarnation.

when working with dynamic musical performances.

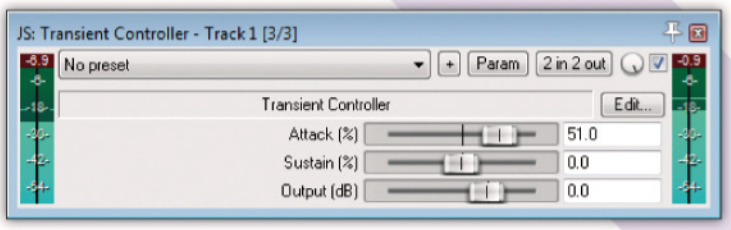
The granddaddy of transient processing was undoubtedly SPL's original 1998 Transient Designer rack unit, and the company continue to manufacture the product to this day (albeit in an updated MkII form), as well as offering a plug-in version. In the software domain, however, transient-processing functionality is now actually built into many DAW systems, and even if it's not available in your recording software, there are plenty of high-quality cross-platform freeware plug-ins available, such as Kilohearts' Transient Shaper, Flux's BitterSweet, or Auburn Sounds Couture.

Practical Applications

But enough of the theory. How can you get the best results out of these dedicated transient processors in practice? Well, the good news is that the main control, often labelled something like Attack, is extremely simple. Turn it up to get more attack. Turn it down to get less attack. And if all you want to do is add a bit of snap to your drum loop, or remove unpleasant spikiness from close-miked hand claps, then you may need do nothing more than just twirl that control and trust your ears. Indeed, for at least half the transient processors I use while mixing, that's really all there is to it!

That said, a big part of the battle is just realising that transient processing might be able to help, because its usefulness extends well beyond obvious drum-enhancement roles. I regularly turn to it for smoothing out things like cymbal stick noise on drum overhead mics, pick noise on acoustic guitars, string slap on upright bass, or clicks and crackle on samples taken from vinyl. I've even used it on lead-vocal and voiceover tracks to combat abrasive 't' and 'k' consonants, or those little snicking noises you get when the singer's mouth's got a bit dry.

Some DAWs already have specialist transient-processing plug-ins built in. For example, Steinberg's Cubase has its Envelope Shaper and Cockos Reaper has JS Transient Controller.



For programmed electronic music, transient reduction can be really useful for smoothing out complex background-pad presets that include percussive step-sequenced or arpeggiated elements that would otherwise distract the listener's attention from more important foreground parts such as a topliner's verse. If you like combining multiple sampled breaks, or underpinning breaks with programmed beats, then it's not unusual for some of your layers to be more important for their timbre and sustain rather than for their attack. In such cases, you can sometimes use transient reduction on those textural layers to reduce any rhythmic flamming between them and the other layers, thereby giving a cleaner and punchier attack to the overall drums submix.

Headroom Matters

From a technical standpoint, though, you should be aware that boosting a sound's attack will often also increase its peak level, especially where you're heavily processing drum tracks. In modern DAW software this needn't be a problem, just so long as you make sure you're not inadvertently clipping your master bus or any of your system's analogue outputs (and thereby unintentionally introducing distortion). However, if you're using any downstream dynamics processors (a compressor on the drums bus, say), a big increase in peak levels can significantly change how they respond.

By way of acknowledgement that headroom can become a concern, some transient processors now include a switchable limiting or clipping section to contain unwanted peak-level overshoots. Now, it might seem a bit daft to first boost the level of a drum's transient, and then immediately limit/clip that level peak. But it's »

Sustain Enhancement

Alongside the Attack control of some transient-enhancement plug-ins you may also see a second knob labelled Sustain, and the implementation of the two is actually quite similar: whereas the Attack control operates on signal levels that are rapidly increasing (ie. transients), the Sustain control operates on signal levels that are rapidly decreasing (ie. decay tails). If you're working with drum samples or drum close mics, such Sustain controls can be remarkably effective, dramatically boosting drum sustain, or (in cut mode) drying up unwanted resonances and room reverb. However, any gain change that's applied will usually then reset after some preset period of time, and while this isn't usually a problem with isolated drum hits, it can begin to sound rather unnatural with more complicated mixed signals where multiple musical components are sustaining across each other. So, personally, if I'm looking for serious sustain enhancement, I'll usually turn to parallel compression instead, as this always seems to sound more musically appropriate to me.

If your DAW software doesn't include transient processing by default, or you'd just like to try out some different algorithms, there's plenty of excellent cross-platform freeware available, such as Kilohearts' Transient Shaper, Auburn Sounds' Couture and Flux's BitterSweet.

» important to realise that limiting and clipping won't completely negate the effect of a transient boost.

For a start, there may be some quieter hits that remain within your available headroom, so won't trigger the clipping/limiting process. But even those hits that do come up against the safety endstop won't lose all the benefits of the transient boost. As I see it, this is because a transient's peak level isn't the only thing the processor changes — it also increases the rate at which the level rises as the transient develops (what I like to call its 'rise time'). So even if a limiter or clipper curtails a boosted transient's peak, the rise time will still be faster than it was, and the drum hit will thus appear to have more attack nonetheless. Mind you, a limiter will typically counteract the subjective effect of a transient boost much more strongly than a clipper will, so you may prefer to use the clipping option if you're after maximum subjective attack and are less concerned with the unavoidable tonal side-effects of the resultant clipping distortion.

Refining The Transient

Another facility you'll see on many transient processors is some way to adjust the shape of the processing envelope. At its simplest, this will just be a single Speed control or time slider, which determines how quickly your desired gain boost/cut kicks in once a transient is detected, and how quickly the gain resets once that transient's passed by. More fully featured plug-ins, however, may have all sorts of rather inscrutable envelope options to choose from. Don't let that intimidate you, though, because whatever settings are on offer you'll usually find that they just give a broad continuum between faster envelope actions (which tend to add a sharper attack 'spike') and slower envelope actions (which typically provide a better sense of low-end weight). Once you recognise that fundamental trade-off, it's pretty straightforward to surf the available settings to find the ones that give you the best balance between 'snap' and 'punch'. If I'm wanting to soften transients rather than boost them, though, then I'll almost always choose the fastest gain-change envelope, simply because I find that it's the sharp onset and upper-spectrum content of overbearing transients that usually make them most unpleasant to listen to, and a faster gain-change envelope better targets those aspects of the sound.

On occasion, however, I do want more control over the tone of my transient processing. For example, I regularly encounter project-studio multitracks where the programmer or tracking engineer has tried (unsuccessfully!) to salvage a gutless kick-drum sound by overemphasising some upper-spectrum attack component — for example the beater 'click' of a close-miked acoustic kick drum. The real remedy for this is to boost the low spectrum of the drum's onset transient, which is something that traditional

Online Resources

As usual, I've provided a selection of audio examples to accompany this article on the *Sound On Sound* website at https://sosm.ag/transient-processing. In addition, I've created a special resources page at www.cambridge-mt.com/transient where I've posted a couple of hands-on video tutorials illustrating the two parallel-processing techniques mentioned in this article. Alongside those, I've also collected together links to a wide range of freeware and payware transient-enhancement plug-ins, in case you're looking to supplement your DAW's existing processors.

For more precise frequency-selective transient processing, you may want to investigate feature-heavy spectral processors such as Oeksound Spiff, Melda MTransientMB, Boz Digital Transgressor, or iZotope Neutron Transient Shaper — or else consider using simpler transient processors in a parallel-processing configuration.

transient processing can't really do, no matter how much you play with its envelope settings. Happily, there are a couple of good workarounds here.

The first is to set up your transient processor as a send effect rather than just inserting it on the kick-drum channel — in other words, to use the plug-in as a parallel processor. In this configuration, you can boost the drum's transient in the effect-return channel, and then EQ that transient-enhanced signal independently before mixing it back in with the unprocessed drum sound. So in this specific kick-drum scenario, I might low-pass filter the transient-processed addition to beef up just the weightiness of the drum's impact.

The second workaround is to get hold of one of the more powerful transient processors that allow you to tailor their processing to different frequency ranges — things like Oeksound Spiff, Boz Digital Transgressor, Melda MTransientMB, or iZotope Neutron Transient Shaper. In our kick-drum example, those tools would let you boost just the low end of detected transients in order to implement a similar outcome.

The down side of the parallel processing setup is that it's only really useful for attack boosts, not attack cuts, and you also have to be careful that phase shifts incurred by your EQ don't cause problematic comb-filtering between the processed and unprocessed signals. (If they do, try a linear-phase EQ instead.) But on the up side, with the parallel setup you can use more easily understood plug-ins — which may also be freeware into the bargain!

Whichever tactic you go for, though, my kick-drum example is only one of the potential applications for frequency-selective transient processing. Indeed, many of the use cases I've already mentioned can be more effectively handled in this way. With acoustic-guitar, for instance, you can usually rein in pick noise much more firmly (without unacceptable processing side-effects) by focusing your transient-reduction processing above 5kHz.

Transient Joy

Once you get a handle on what this nifty processing option has to offer, you'll likely find yourself getting mix results quicker, without having to push your EQ and compression settings to extremes — so it's definitely worth spending some time experimenting with the tricks I've mentioned here. One word of caution, though: transient boosts in particular can be strangely addictive, because they'll make most instruments sound more upfront and engaging. But if you make everything upfront and engaging, then your mix will likely sound cluttered and lacking in any sense of depth. As with many things in life, moderation is a virtue.

FLIPPING PATCHBAYS!

Many affordable patchbays come half-normalled by default, and to change that you must flip the cards inside. It's easy enough, but there's a knack...

MATT HOUGHTON

Patchbays are the unsung heroes of the recording studio, and even in a modern home-studio setup they can be worth their weight in... well, maybe not gold but you get the idea! If properly set up, they not only allow you to connect anything in your studio (synths, processors, effects and more besides) to anything else by simply plugging cables into the front, but also to establish a 'normal' routing that works even with no patch cables plugged in.

If you want a more detailed look at planning and configuring your patchbay, check out Hugh Robjohns' excellent article, Patchbays In The Modern Studio (*SOS* December 2020: www.soundonsound.com/techniques/patchbays-modern-studio). He runs through the differences between normalled, half-normalled and isolated (un-normalled) setups and their pros and cons, as well as describing some useful utilities such as phase inverters and mults, which you might want to include alongside your gear. In this article, though, I want to share some simple, practical tips for setting up affordable TRS jack patchbays.

Overview

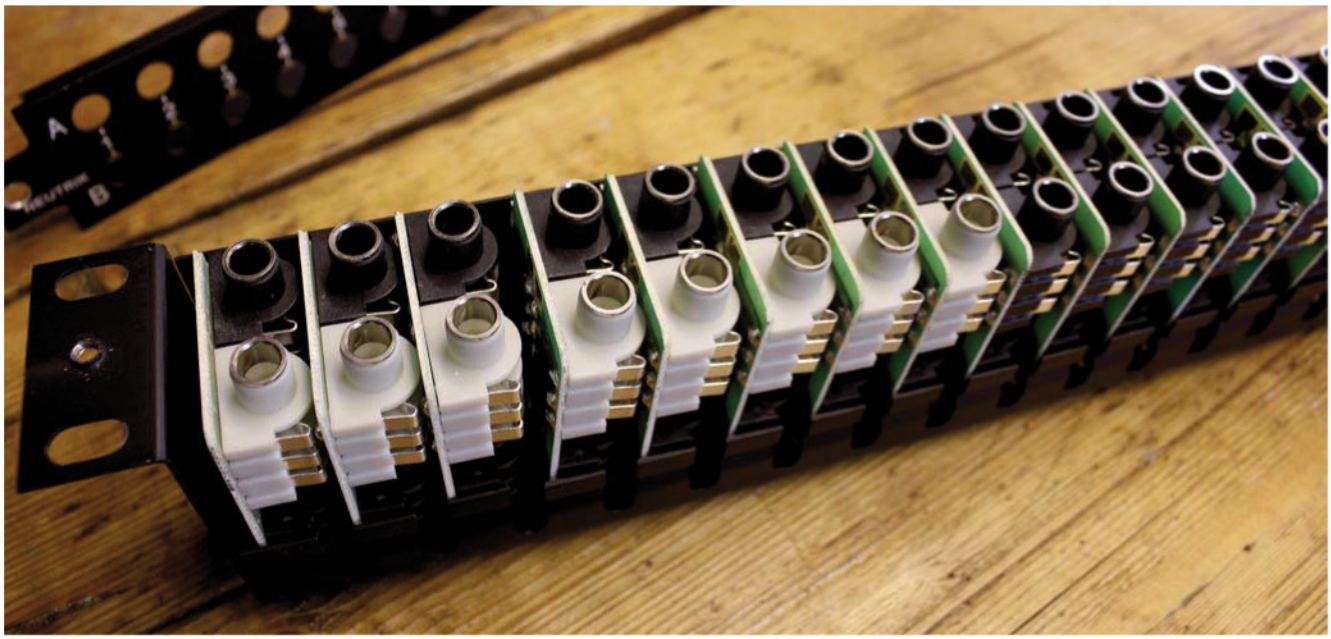
I've long used patchbays, but it's only recently, when setting up a new system, that I've used Neutrik's affordable NYS-SPP-L1 patchbays; shop around and these can be picked up new for under $100 each, and you might even find similarly designed models under different brand names slightly cheaper. While I can't vouch for those, I can tell you that as long as these Neutrik ones are used with reasonable care, they're perfectly decent and robust enough for a typical home/project studio.

> "You need some sort of 'poking device'. And the perfect tool for the job is the flat-head screwdriver that you'll have to hand already."

Some slightly pricier patchbays, still at the budget end of the market, feature convenient switches to select for each channel (a vertical pair of jack sockets on the front, and a corresponding pair on the rear) whether the connection is normalled, half-normalled, parallel or isolated. Samson's S-Patch+ and Behringer's PX3000 spring to mind. I prefer the former, where the toggle switches are on the front, over the latter's top-mounted slide switches, which are inaccessible once installed in a rack and can sometimes slip out of position if you need to slide the device in/out of a single rack space. But really, if you plan your patchbay properly, the normalling is not the sort of thing you'll need to change on a daily basis.

The Neutrik patchbays I installed recently (and others that adhere to the same basic design) may be slightly less versatile, but they do still allow you to change the configuration for each channel. To change from the default half-normalled setup to a split or isolated one, all you have to do in theory is pop the front plate off the patchbay, pull out the channel module (a PCB with the four jack sockets attached), and flip it around. One jack socket has a grey collar, and when this faces forward it's half-normalled. When it's at the back of the patchbay, it's a split configuration, whereby the front two and bottom rear jacks are connected. In that scenario, when a jack is inserted into the grey socket at the rear (which obviously happens when you hook your gear up to it!) the top and bottom front-rear pairs become completely isolated from each other.

It's a neat system in theory, since it means you can mix and match the half-normalled setup for your most commonly used signal paths and an isolated one for things you wish to patch in more occasionally. By connecting patch cords around the back, you can even create a passive splitter (ie. a parallel/mult path), whereby you plug a patch cord into the top front jack of a half-normalled channel, and



Inside the patchbay, each 'channel' comes in the form of a card fitted with four TRS jack sockets. These can be popped and pulled out, allowing you to flip the cards around to change the routing — the position of the pale grey jack casing (the other three have a black casing), which can poke through the front or rear panel, indicates the routing arrangement.

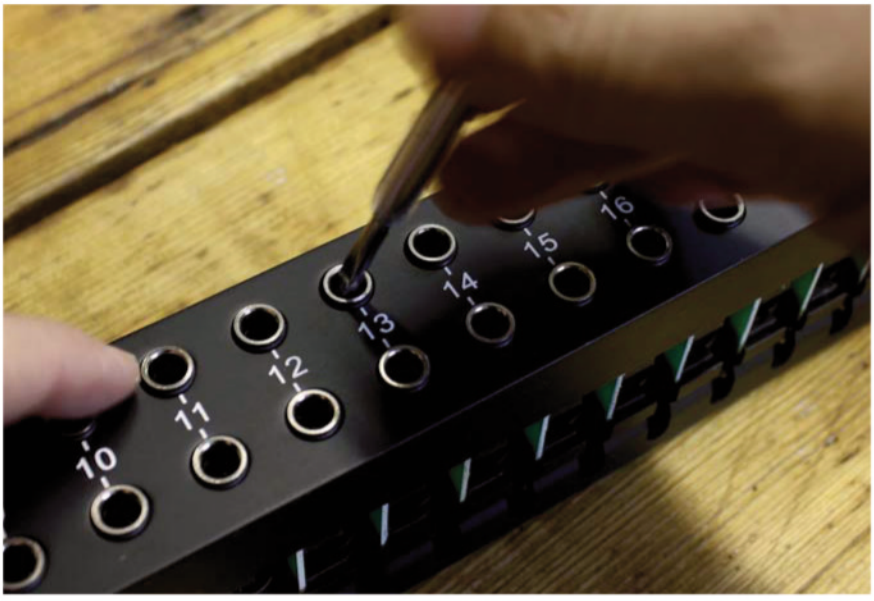
link the two rear jacks to the corresponding ones on an adjacent split/isolated pair. There's also a separate replacement card available that caters for insert points, which allows you to connect a mixer's (or perhaps a preamp's) unbalanced insert sends/returns to the patchbay using regular TRS to TRS looms, rather than requiring Y-cables.

Practicalities

I said it's neat 'in theory' because I've seen reports online of this style of patchbay being rather flimsy. These cards are all held in place by the patchbay's front panel, which has holes drilled to allow the jack sockets to poke through just enough to be identified (grey or black, remember?). The Neutrik model has a pretty solid panel, which exhibits only a little flex once it's been removed (I can't vouch for the solidity of other brands) and practically none when it's screwed on to the chassis.

Still, it's possible to mess things up. First the cards are a snug fit, and to pop them out takes some force — that is, until you discover that you can simply press in the jack sockets at the rear to unseat them! Second, and more importantly, when reseating the cards you have to make sure that the jack sockets poke neatly through the holes. I'm pretty sure that some people aren't getting the cards to sit quite right, so although the front panel manages to retain the reoriented cards, it exerts less than the intended pressure on them. Thus the unit as a whole starts to flex more and feel rather cheap when you start inserting your patch cords.

To avoid this, all you need is some sort of 'poking device' that you can insert into the top jack sockets — just use this to manoeuvre the cards into place as you reseat the front panel. And the perfect tool for the job is the flat-head screwdriver that you'll have to hand already because you need one to undo/tighten the two screws that hold the panel in place! Do this methodically, working from one side of the bay to the other, and you'll find that the cards soon fall into alignment with a minimum of fuss and pretty much zero need to fiddle with the positioning of the front panel itself. The only other thing to bear in mind is that, once these are properly aligned, you might find that you need to press on the top of the sockets to reseat them properly — this both ensures that the sockets protrude as they should at the rear, and that the front panel is a good, snug fit.



When reassembling, it's important to seat the front panel correctly so that it fits over the cards snugly, and keeps the whole affair feeling robust and rigid. The 'trick' is to seat the panel over the cards, and then use something like a screwdriver to move the jacks into place.

Euro Voices

Vocal Mixing: Lessons From Eurovision

A broadcaster's personal guide to getting a great vocal sound at Eurovision. Or in a pub.

ROBERT EDWARDS

Over three live broadcast shows, with 37 different songs from 37 different countries, the Eurovision Song Contest has become a unique event that brings together enormous technical resources. Over 500 million viewers tune in to watch an intense competition between the sovereign states of Europe, Israel and Australia. No-one is harmed in the process, apart from the occasional dent in national pride of those getting fewer votes than the UK. Every country's entrant wants the live broadcast to capture their unique sound, and that sound must match the publicly released commercial download version on Spotify, TikTok, YouTube or Apple Music. A simple enough task on paper.

Each country's entry is chosen by the middle of March each year. This gives a couple of months for every department to get their act together to plan how they will manage all the varying requests when it comes to the live shows in May. They include sound, cameras, lighting, screens, costume, set design, logistics, power, infrastructure, accreditation, engineering, make-up, floor management, security, catering, IT... the list goes on. And that's not to mention the requirements of 37 different commentaries on-site, or the 37 incoming voting feeds from the EBU.

Holding The Line

From the very first Eurovision held in 1956, through the UK's Sandy Shaw and her winning song 'Puppet On A String' in 1967 (ask your mum and dad), right up to Lisbon Eurovision in 2018, all the singing, including the backing vocals, had to be performed live.

Things have changed since Covid arrived. Eurovision now allows quite a juicy backing track, and backing vocals can now be on the playback. However, competition rules prohibit 'lead dubs' where the track offers

An artist's impression of the Liverpool 2023 arena (the real thing was not finished at the time of going to press!).

too much help for the live lead vocal. It must not appear to be mimed, and the live nature of the singing is further respected by the rule that no form of auto-tune is permitted.

The overriding challenge from an engineering point of view is to 'make it sound like the record' in a live context, for TV and radio broadcasts. There are also further challenges thanks to the very restricted and tightly controlled rehearsal and transmission schedule. As it really is a competition, the rehearsal schedule runs in 30-minute blocks, one per delegation, to ensure fairness is achieved for each country. There are also just 40, count them, seconds to accomplish a complete changeover from one song to the next during the Postcard films. Artists, microphones and complete sets of scenery are all swapped by a massive army of highly skilled technicians and stagehands. What could possibly go wrong?

Robert Edwards at Abbey Road Studios, where the tracks and stems supplied by Eurovision delegations are checked to see whether they meet the show's technical requirements.

Be Prepared

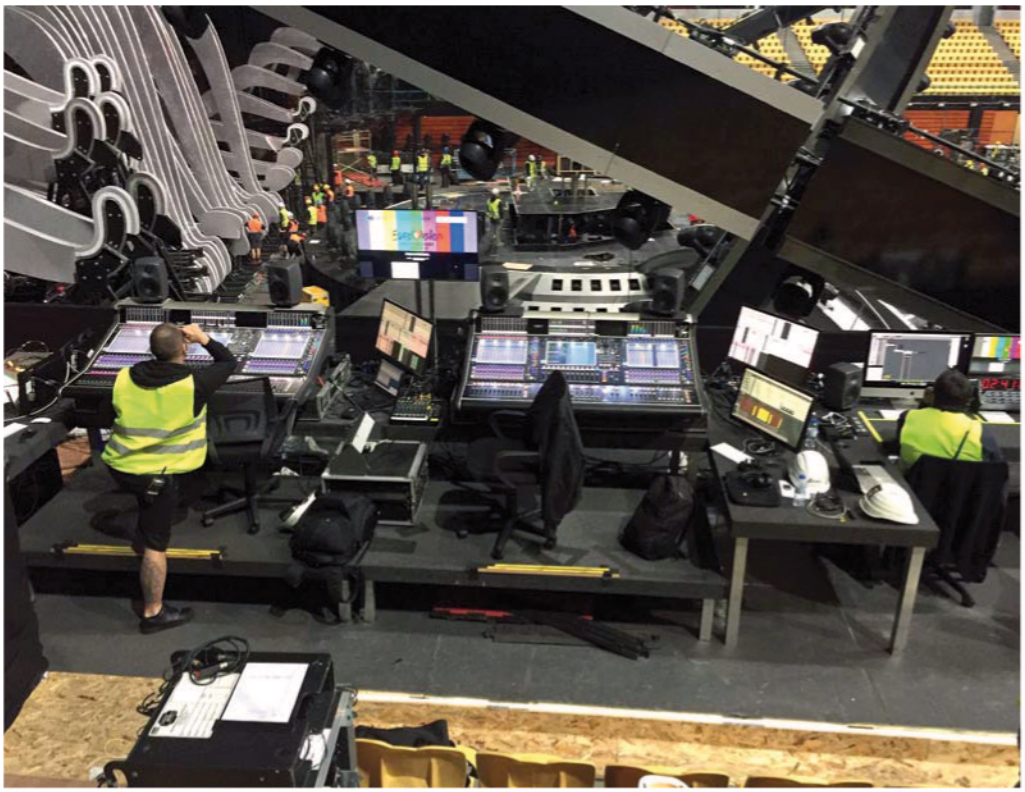
So, what is the secret in getting a great vocal to 500 million radio and TV listeners whilst adhering to such a tight schedule?

It obviously all begins with the singer, the song and some serious planning. No-one should turn up to a gig in a pub or world event without doing some proper homework. What stylistic vibe are they trying to portray? Jazz, country, house, R&B, heavy metal, rock, pop, garage, bike shed? (Maybe the last one is still too niche.)

You need to know how serious the sound is to them. Do they have their own IEM (in-ear monitoring) moulded earpieces and a system? If not, should you supply them? Do they use monitor wedges, or just rely on the in-house pub or stage PA? Do they have a proper vocal warm-up routine, or do they just gargle straight vodka?

Any lead singer's standard response for a poor vocal performance is always to pin the blame firmly on the IEM mixer or the PA system. This the number one standard response in the vocalist's guidebook of excuses when the performance is out of tune and/or out of time. Get over it.

At Eurovision, the delegations are obliged to use IEMs, as the stage area and arena is so huge. That has a positive effect on the sound, as unwanted spill from loudspeakers into the vocal microphones is vastly reduced, which helps to achieve the goal of achieving happy delegations and a better chance of getting that 'record' mix. Using IEMs reduces coloration, but the PA design and operation also plays a vital part. Keeping 30,000 people in »



This backstage shot shows the monitor position from the 2018 Lisbon contest.

» party-mode for three hours in an arena without alcohol takes a lot of Kilowatts. It's arguably easier in a pub gig with free-flowing lager.

There are hidden wedges built into the Eurovision stage to provide audio energy, and to provide a healthy backup if the IEMs ever fail or drop from a performer's costume while doing a headstand during the act, and there are also large slung sidefills to help the dancers.

Head Or Hand

Good planning will involve choosing the right microphone. You can't go wrong with an SM58 on a cable in the pub, or by aiming a U87 at virtually anything that makes a sound in a studio. But there is a lot of smoke and mirrors regarding the matching of a microphone and capsule to suit an artist's voice. There is a brotherhood of monitor engineers flying the globe in Club Class, each with a flightcase containing just the perfect combination of microphone and capsule for their headline artist. In truth, the differences between them all are generally quite small, and these monitor engineers are there more to get the IEM mix right, and to provide encouraging and enthusiastic support to the artist in the hotel bar afterwards.

To achieve a level playing field at Eurovision, all the artists from all the delegations are obliged to choose from our deliberately limited range of radio microphones. It's the Henry Ford choice: hand microphone or head microphone, black or black. It keeps it fair.

Hand mics generally sound better, but for all sorts of reasons, generally based around costumes, staging and styling, head mics are sometimes chosen by delegations. There is a minefield of potential issues in using head mics, including earrings that jangle like the contents of plumber's toolbag, hairspray applied to the hair and copiously into the mic capsule just before going on stage, or simply that the costume has nowhere to mount any of the IEM or mic packs. Any solution must involve the ability to easily change batteries or to adjust the levels on the actual packs. Fortunately, battery management revolves around rechargeable technology these days, and the remaining battery indicators for most packs are uncannily accurate, unlike those on most electric cars.

This year, Shure and their Axient system fitted with DPA capsules are the preferred choice of radio microphone, but Sennheiser have also been main providers in the recent past. Frequency management is one of the major challenges, and almost all the available UHF spectrum from TV channel 21 up to TV channel 49 is required to accommodate all the digital radio microphones, IEMs and communication devices needed on site. OFCOM, the government licensing body, are very involved in the process, and are on site to keep a watchful eye over any music that might appear from any unexpected Liverpool bedroom transmissions during the event. Expected transmissions from Liverpool include the BBC, who have Radio One and Two broadcasting live during the week, and a daily presence from *The One Show* on BBC1, all requiring their own chunk of the spectrum. Marconi would be impressed.

The Fundamentals

The principles of getting a great live vocal sound are the same regardless of the »

Backing Tracks & Backing Stems

For a normal TV appearance, most artists present a stereo backing track with a separate click stem for the IEM, but Eurovision asks for much more. The countries must deliver a raft of stems that together form the backing track, including separate backing vocals. This is to satisfy the need to generate mixes for TV, the IEMs and the arena audience hearing the front-of-house PA system.

The click track does what it says: as a hidden track, it provides a tempo click just on the IEM feed. It can also include a vocal count-in, and maybe a tuning note before the start. During the intro of an a cappella number, it often contains a ghost piano for the artist to sing with, again only appearing in the IEM. This method is often featured on talent shows to provide a killer start.

But nobody can make a killer vocal sound if the backing track mix is so dense that the sound of a blue-lighted ambulance would struggle to cut through. The backing track should be just that: a track to support, to reflect, and to wrap the singer. That doesn't mean it can't be loud, but it shouldn't have the vocal stuck on top like a cherry unless that's the wanted style. If you haven't made the backing track yourself, put the musical director on your speed-dial list.

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» scale of the event, and they start with the source. Talk to the singer. Make sure they hold the mic properly, at a reasonably constant but close distance. Also, get them to understand that if they strangle the capsule by holding the mic at the top, the frequency response will become very unpredictable, likely causing feedback, or at worst, it will sound as if they are singing into an empty Heinz can. Fit an appropriate windshield if needed.

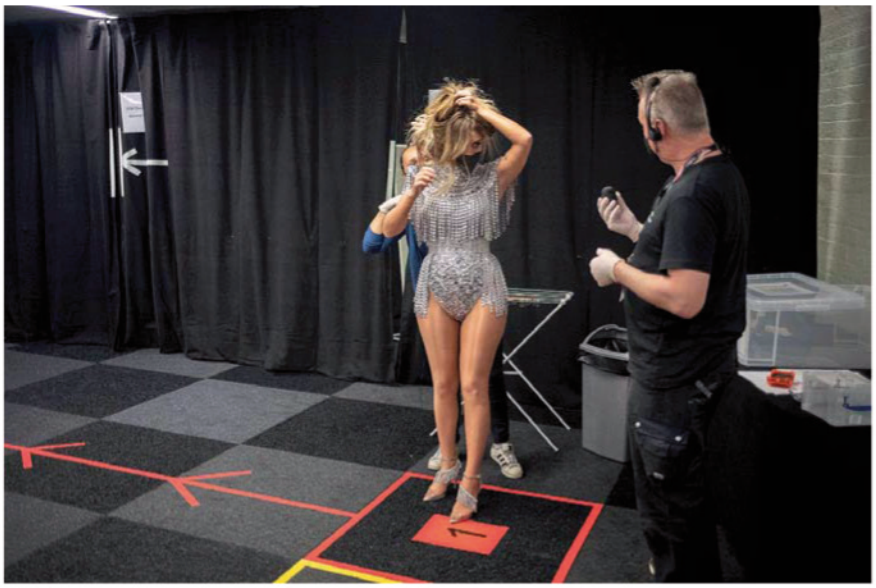
The next job is to sort out the microphone input gain, whether it's a radio mic system or on a cable. Shout or scream into the device to make sure you set the gain below any clipping, then reduce further to give yourself some more dB of headroom. If the signal is distorted at this stage, you have no hope. And remember, by definition, there are no bits to describe a level above 0dBFS, so if your digital levels are very high in the quiet parts, you are already on a path to ruin. There's absolutely nowhere to hide when the lungs expand for the chorus and your mix literally flatlines.

The radio microphone sound generally arrives digitally via fibre back to the broadcast truck. So what happens next? We listen. We check it sounds 'clean' from system noise, hum, or digital clicks. Then we start filtering. We get rid of anything that is either annoying or we can't hear, so that means anything much below around 100Hz, which is generally just unwanted thumps, pops and bumps. We also get rid of anything above 19kHz, as nobody will really notice, and it can stop unexpected aliasing effects further down the chain. If you can't hear it, get rid of it.

In The Box

The mic signal next takes a trip through a compressor. Compressors get a bad press from the purists, but it's unfair to blame them when they can do so much good if used in the right way. Really get to know your compressor, and get comfortable with every parameter and characteristic. Try to stick with a few of your standard settings from one or two different manufacturers. Like a friendly dog, take it on every gig with you.

While on the subject of our canine friends, just think of a letterbox. In broadcasting, we must post the audio through a fairly narrow loudness slot defined by R128, the EBU delivery standard, set at -23 LUFS, so that it satisfies the audio delivery police. If it's too loud, we can't get it through the letterbox without seriously damaging the content. If it's too quiet, no-one hears it drop on the mat and the viewers turn over to a louder channel or send letters to the *Daily Mail*.



Eurovision costumes are not always designed with battery packs and cables in mind! (Photo: EBU / Andres Putting)

There are no rules about how much compression to use, but a commercial vocal will generally sound better with some good compression. Start with 3.5:1 ratio with a 4ms attack, followed by a 50:1 limiter spaced some way above it with a faster attack. Begin there and then make your own judgement.

Then comes the magic of equalisation, although some prefer to do that before compression. Clean out any remaining dross below 100Hz, scoop out a bit at 500Hz, and brighten up the octave between 5 and 10 kHz. You can fiddle for ages in the mids, but maybe save that work for once you've got the track level sitting exactly where you want it.

Now for the polish (not the Polish, who haven't yet won the competition). Natural reverb and delays have been around since the dawn of time, and the ear loves to hear a human vocal in an interesting acoustic space. Caveman loved a good slapback echo, monks love to sing in a reverb longer than five seconds. Gen Z now has an incredible range of toys that can excite the ear, but still most modern vocals predominantly rely on a combination of caveman and monk, using some delays and some reverb.

Typically, there will be some repeated delays in time with the crotchet beat, coupled with a shortish reverb. For a ballad, try a long pre-delay of 120ms coupled with a longer ambient reverb of, say, 2.6 seconds. For Eurovision, there is intense listening to and copying of the effects used on the 'commercial release' version, which might contain a smorgasbord of phasing, repeats and reverse delays.

Learn to work quickly and confidently by reducing your equipment choices to the kit you really know well. This applies especially to the reverb and delay devices you choose. It certainly applies to both 'down the pub' and Eurovision. There are very few artists who ever want to hear their vocal completely dry.

Keep Calm & Carry On

The golden rule with vocal mixing is always to remember the basics. Don't over-complicate. Be kind to the stressed-out artists, and eventually they might say something nice to you about your mix, but don't bank on it. Remember, if you are using a battery-powered device — say a laptop with those brilliant compressor and delay plug-ins, or maybe a radio mic — it will always try its best to run out halfway through a song. Be cunning and wise and have a backup. Finally, remember a roll of tape and a pen to label and identify everything. In the pub, label the 13A plugs so you switch off the jukebox not the PA. This practice will hold you in good stead if you ever have the joy of simultaneously wrangling 120 radio devices. No-one should ever have to play microphone roulette with 500 million people listening in.

Robert Edwards FIPS is a BAFTA Award-winning Freelance Sound Director who was Head of Sound for Eurovision 2023 in Liverpool.

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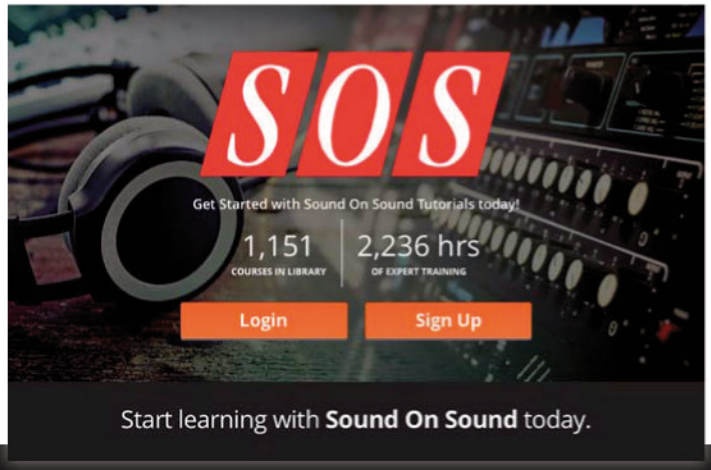

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SOUND ON SOUND

We take a look at what's new in 6.1.

ROBIN VINCENT

PreSonus recently updated Studio One to version 6.1. It's the first significant update to version 6 and brings with it a reasonable bunch of improvements for the Song page, Project page and Show page. There's no startling new functionality, but there are a number of solid tweaks that are going to, in DAW developer language, enhance your workflow.

Colouring In

Let's go straight for the most marvellous workflow enhancement of all, which is that you can now use your own palette of colours on tracks and clips. Previously, when you clicked on the colour bar on a track, clip or mixer channel, you were offered a choice of 100 or so colours curated by PreSonus. Now you can expand the colour-choosing window to reveal a colour wheel. You can create your own palettes by selecting a range of colours defined by HSL (hue, saturation, lightness), RGB (red, green, blue), hex values, or just by clicking on it.

It's precisely the sort of distraction we need right now, and PreSonus have even generated some preset options for you to consider. This feature gives you the ability to create a palette with just a handful of colours, making it much quicker to choose one. You can also share your custom palettes in the PreSonus Exchange hub, so other users can enjoy your damn fine taste.

Timecode Display

Studio One already had a timecode display, but this lived in its own independent window. Version 6.0 brought in a lot of really nice video editing features, but the timecode display and video preview still occupied different windows. Now you can have the timecode running in the bottom corner of the video display so you no longer lose track of it.

Another video tweak is that you can import video from within a Smart Template. When you choose a template from the Studio One Start page, you can drag audio tracks to the Drop Zone. You can now to the same with video files too, so that your project starts up with everything you need for your cinematic pop video.

Sphere Notifications

PreSonus have been taking steps to further integrate what happens in Sphere with your working DAW environment, and if you are a sociable music producer and have enjoyed collaboration using the workspaces in PreSonus Sphere, you will be thrilled to know that you can now get notifications from your collaborators directly within Studio One. Previously you'd have to access those sorts of communications separately, either in a browser or the MyPreSonus App. Now you can get yourself interrupted right in the middle of a session.

Anything that happens in your cloud-based workspaces can send out notifications. And as soon as you've exported a project to the cloud, all your collaborators will know about it without having to check their email.

Anything that speeds up the way you can connect and collaborate has to be a good thing. There's always the potential for issues when trying to do something in software that requires a default browser to pop up and facilitate things. This integration works brilliantly and bypasses the need for running email or other software on your music production system.

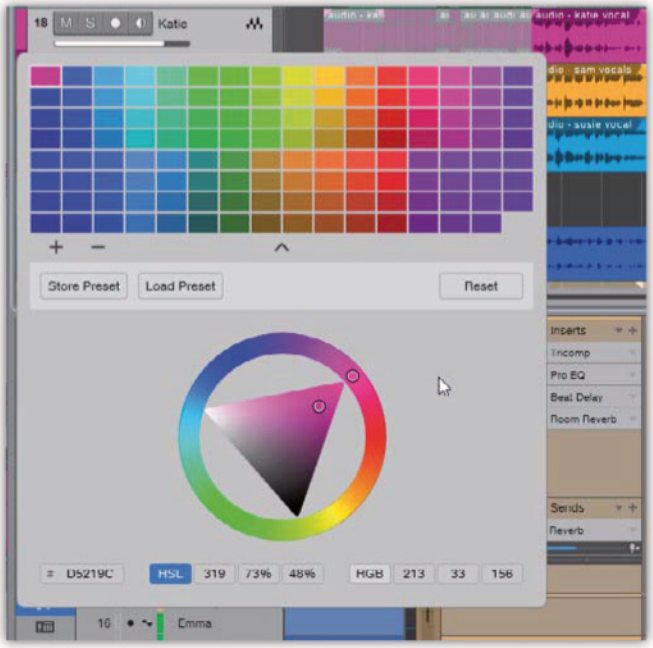
You can also turn off notifications in your Sphere preferences.

Sync Points

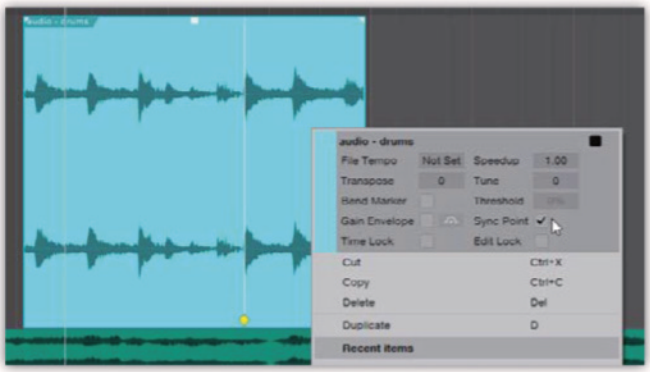
Arguably the coolest development in the 6.1 update is a new timing feature called Event Sync Points. If you have unquantised audio within a container clip on a track and you want to duplicate it to another place in your song, it can be tricky to place it accurately. As the clip is not aligned to a grid, you're going to have to line it up manually when you move it to another spot. It's not that it is difficult to do, but it can be very time-consuming. Not any more.

There's a new timing reference you can add to your audio. It's called a Sync Point, and it's accessible from the right-click menu. Once enabled, a small yellow diamond appears in the bottom-left corner of the clip. You can grab it with your mouse and set it to a specific position within the clip. This becomes the Sync Point around which the audio syncs and snaps to the project. As you move the sync line it snaps to the peaks and transients of the audio, making it really easy to find the ideal point. So now, wherever you move that audio clip to it will snap its Sync Point to the grid.

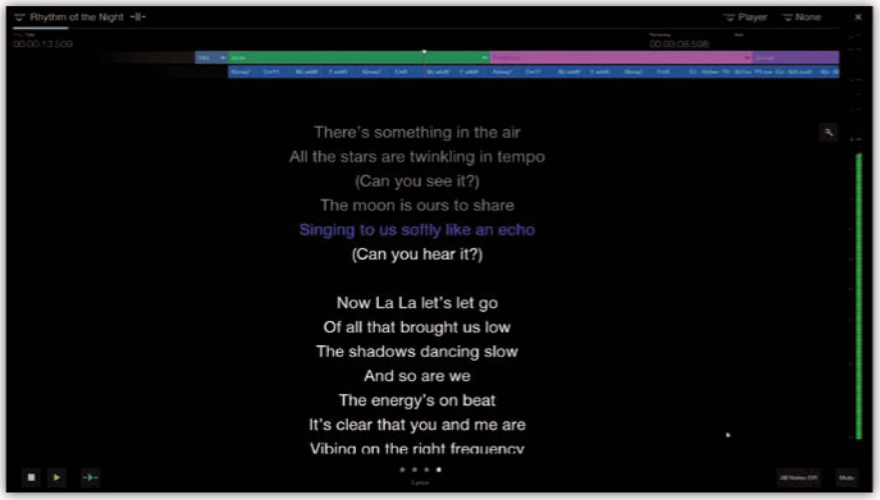
This is also useful in a free-flowing project that's not using a grid. You could set Sync Points on various clips on different tracks and then enable the grid momentarily to line them all up. The audio is



You can now choose custom colours for tracks and clips, and even make — and share — your own colour palettes.



Sync Points help you line up your audio clips, even in projects that don't use the grid.



You can now edit lyrics from the Show page.

then sync'ed to transients rather than the clip containers. You need to enable Snap to Events under the Snap menu to make this happen. I think this is a huge time-saver.

Lyrical Updates

The Lyrics engine was one of the best new features of version 6. In the Song page, separate lyrics are now available for the Scratch Pads, so you can work on alternative words in the Scratch Pads as well as different musical ideas. The lyric display switches depending on what you have in focus. Editing has also been made a lot easier with a ripple function: if you pull something out or decide to move things around, the rest of the lyrics shift to accommodate that without you having to re-edit everything.

You may have noticed how the lyrics get truncated in the lyrics lane, especially when you zoom out and there's no room to keep them in view. You can now hover your mouse over any section and it will display the full lyric. Lyrics can also be scraped out of MIDI files, so if you are importing a MIDI song file, your life just got a whole lot easier.

While it's a tremendous songwriting tool, it's also a brilliant thing to have in the Show page, so that you can display and follow the lyrics during a performance. The thing was that the Show page only used the global view pulled in from the Song page and not the more comprehensive lyrics lanes. Well, guess what? Now it does. The integration is much better; you can edit directly and rather than popping up in another window, the Lyrics view is part of the view choices and has the chord arrangement running along the top.

Finally for lyrics, you can now display them in the Studio One Remote App for iOS, Windows or Android. The app has been updated to version 1.7 to support all of Studio One 6.1's features.

Plug-ins & Templates

While I continue to bang on about the lack of new instruments, PreSonus do put some energy into the effects plug-ins. The Pro EQ now has some very useful solo buttons on all the shelf filters, so you can isolate individual bands that you are working on to get down to some serious detail.

There's also been a tweak to the new Vocoder. PreSonus have added a pink noise option for a bit more flavour.

We enjoyed the arrival of Smart Templates in version 6. This included a single template for the Project page that opened the default project. Unlike the Song page there was no way to create alternatives or User Templates. Now the template function has been ported over to the Project page so you no longer have to be faced with the emptiness of the default.

To create a template, add in the plug-ins you always use, select the metering you prefer and lay the windows out however you usually do. Then go the File menu, choose 'Save as Template' and the job is done. They appear as User Templates under the New button on the Start page. I imagine that's saved a lot of clicking about or confusion when opening an older project to re-save as a new one to use the same settings, which is what I always do.

The other new function is that Loudness now has an Album mode. This is within the Digital Release section, where you can adjust the target loudness of your exported files. This was done previously on a song-by-song basis whereas now, with the Album mode, it will consider the overall volume of all the tracks together. This will remove any unexpected volume changes between songs.

Finally, In a couple of months I'll have written 50 Studio One workshops. While there doesn't appear to be an end to the functions within Studio One we can talk about, I would be very interested in hearing suggestions on what you'd like to see covered. Please send your requests to feedback@soundonsound.com for consideration.

The Pro EQ plug-in now lets you solo your shelving bands, as well as the parametric ones.

Screen 1: Drift's easy-to-follow control panel.

Float away with Live 11.3's new synth...

SIMON SHERBOURNE

Live version 11.3 has arrived in public beta form, and it's MPE-themed. But don't switch off if MPE is not your thing or you don't have an MPE controller. There's a brand-new instrument device that we're going to explore here, and although it supports MPE it works just fine without it. We'll dig deep into MPE and its integration in Live 11 another time.

Drift features a familiar two-oscillator subtractive synth engine, and I guess was conceived as a new default virtual analogue sound source in Live. I say this because it will be in all versions of Live from Intro upwards, and has a simple user panel that's more approachable than the Analog device. It's less dense in its layout and has Push-friendly graphics. It's not just a cut-down Analog, though. It is slightly less flexible in its routing and has fewer onboard mod sources, but its oscillators are considerably more versatile and it has a whole bunch of exciting and unexpected stuff going on that might not be apparent at first glance, making this a really special little synth.

Shape Shifting

The first block on Drift's panel accesses two oscillators and a noise source. Each oscillator has a pop-up shape/mode selector, with osc 1 having a few more options. Ableton say that Drift is influenced by modern synths and modules, and you certainly feel that here. Osc 1 has a Shape control that appears to do different things depending on the basic oscillator mode. There's an oscilloscope display that shows the product of the whole oscillator section to give clues as to how the sounds are being manipulated.

With the sine and triangle shapes, the Shape control appears to apply wave folding. The third mode provides morphing between triangle and saw. The fourth shape looks like half a sine and triangle stitched together, which the Shape control progressively clips, providing a pleasing harmonic sweep — a great starting point for classic poly sounds. The sawtooth and pulse modes get a sync sweep treatment from the Shape control, while the final square mode gets pulse-width modulation.

Note that slap in the middle of the oscillator 1 section there's a modulation assignment mapping MPE Slide to Shape. If you look across the other sections you'll see that most have these local mod assignment options, allowing you to make quick patches in addition to any modulation you set up in the main mod sub-page. If you don't have an MPE controller available, you can flip the Shape modifier to an envelope, velocity, mod wheel, etc. At the bottom left there are two further mod selectors that affect both oscillators. Usefully these default to env 2 (to quickly dial in kick or 808 bass-type sounds), and LFO (bring on the Boards Of Canada wonkiness).

Dynamic Filtering

The Osc Mix section has a nicely intuitive interface, with on/off buttons for each

Screen 2: Each of the oscillator modes are affected differently by the Shape control.

oscillator, level dials, and arrow buttons that set whether each source routes through the filter or bypasses it. The primary low-pass filter has two modes. One is 12dB/octave and one is 24, but they are also tonally different analogue models. This is complemented by a high-pass filter. Again, there are a couple of direct mod source selectors, with MPE Pressure pre-mapped. I've made a habit of switching this to env 2.

One of the many dashes of secret sauce in Drift is the dynamic interplay between the oscillators and filter. Firstly there's a progressive and pronounced saturation that comes in even at modest input levels. The oscillator levels default to -6; anything above that will start to transform the sound. As both oscillator gains are modulation destinations you can use this to add harmonic movement.

The resonance of the filter also responds differently to different input levels (and between the two modes). At very high levels the resonance has little effect on the sound, but it gets more pronounced as you back off the input. Then, curiously, as the input from the oscillators drops to -12dB and below the filter starts to self-oscillate. If you set the filter to key-track you can effectively crossfade between your oscillator sources and tuned-filter oscillation by adjusting the levels.

Mod Tricks

Next up we have two envelopes and an LFO. The first envelope is a fairly standard ADSR, with exponential decay and release. The second is the same unless you put it into looping mode. This essentially turns env 2 into an LFO or function generator that resets on note triggers. The controls change from ADSR to Tilt, Hold and Ratio. Tilt bends the shape of the cycling envelope from its default triangle toward a ramp wave in either direction. At the extremes this produces an exponentially curved shape. Hold adds a time stage between the rise and fall, and Ratio adjusts the frequency.

Screen 3: Both the LFO and cycling envelope can be sync'ed to the audio-rate oscillators and used for FM.

This is where things get even more interesting. Both the LFO and cycling envelope can be manually set to Hz or ms speed references, but can also be sync'ed to tempo or to the oscillators. This last option opens up FM as a sound-design option, letting you modulate the pitch of the main oscillators. Adjusting the shape of the envelope creates shifting timbres. The LFO has several different shapes to choose from for different-sounding FM results. The sample and hold option introduces noise of different flavours.

The LFO has a local Amount control which adjusts its amplitude, and this can be modulated. This gives you a way to scale the amount of LFO modulation over time using the envelopes (or other expression controls) so you can make traditional FM sound-design choices such as adding an aggressive timbral attack to a sound.

Guided By Voices

By now you'll probably not be surprised to hear that the humble Voice Mode section of Drift holds a lot more goodies than a run-of-the-mill synth. Four voice modes are available: Poly, Mono, Stereo and Unison. Choose anything other than Poly and an additional Amount control appears. This is self explanatory for Unison mode, applying the classic stacked, detuned voices. In Stereo mode, voices are panned across the stereo field for a wide sound. But the surprising option is Mono. With the Amount control at zero percent this provides a regular monosynth voicing. However, as you increase the Amount, additional sub-oscillators get blended in for extra fatness.

Last but not least we reach the control that gives the synth its name: Drift. As you can probably guess, this control introduces pleasing analogue-style instability, altering the pitch and tone of each voice. This interacts in juicy ways with what's happening within the voice modes, and is particularly gorgeous in stereo spread mode.

More MPE

Drift may be the new MPE showcase instrument in Live, but 11.3 also expands on the MPE capabilities of various other devices and areas. The existing instruments Analog, Collision, Tension and Electric have been added to the list of MPE-aware devices (which was previously Wavetable, Simpler/Sampler and Arpeggiator), and you'll find Pressure and Slide controls added to their user interfaces. The factory patch library has been updated to take advantage of these new abilities. The MPE Control device, which lets you manage and shape incoming MPE and convert it to traditional performance modulation, has had some refinements, including the ability to convert Pressure to Slide for flexibility when controlling a synth that can use aftertouch but not MPE. An interesting new trick is MPE in the Note Echo MIDI effect, allowing you to map all three flavours of MPE movement to the feedback control, which is fun.

Screen 4: Like many of Live's devices, Note Echo now has dedicated MPE controller support, in this case to dynamically control feedback.

Is Pro Tools still king when it comes to editing audio?

JULIAN RODGERS

All the major DAWs are more than adequate for typical music production tasks, but it is true to say that the perception of them varies, and each is seen to have its own particular strengths. Of course, to make a straight comparison between DAW features you have to know each DAW equally well, something not many people can really claim. But it's a common opinion that Pro Tools is particularly well equipped for audio editing and mixing. Other DAWs may have equivalent features, but something about their implementation in Pro Tools makes it as fast, if not faster, than anything else out there. Why is this? Here are five features which really speed up audio editing in Pro Tools.

Command Focus Shortcuts

Pro Tools has two sets of keyboard shortcuts. The first are the conventional keystrokes, which involve use of a key with one of more modifier keys to invoke a function without having to click on any menu items or icons or drag anything anywhere. Examples of this category of keystrokes would be things like Command+M to mute a clip or Option+Shift+3 to consolidate a selection.

All DAWs have keyboard shortcuts, but something which makes Pro Tools particularly fast is the second set of keystrokes which are mostly based around single keys. These Command Focus keystrokes can be switched on and off, either by using the Command+Option+Shift+1 shortcut or by clicking the little 'a-z' symbol in the top right of the Edit window. The reason it's possible to switch them off is so that you can use one of the two other keyboard focus modes, whereby Command Focus can be used to select and deselect Mix and Edit Groups or to control the clips list. A good tip is to use the corresponding shortcuts of Command+Option+Shift+2 and Command+Option+Shift+3 to access these other modes.

Another historical reason why Pro Tools has enjoyed the status it has as a fast editor is that, until Pro Tools 2022.4, it wasn't possible to re-map the default keystrokes. This meant that an experienced user could take it for granted that any Pro Tools system they encountered would have identical key commands to hand. It is now possible to customise shortcuts by visiting the Keyboard Shortcuts window (Setup / Keyboard Shortcuts), which also lets you check the current setup of your shortcuts, and restore them to the default if desired. If you want to check out the available Command Focus keystrokes you can filter them using the Focus button at the top to peruse the complete list.

To see the complete set of Command Focus shortcuts, use the Focus filter button in the Keyboard Shortcuts window.

Tab Fab

Being able to quickly select the audio on which you want to perform an edit is essential if you want to work quickly. You can make selections visually using the Selector tool, or by ear using the up and down arrows during playback. Using the arrow keys is an overlooked way of making selections but is really useful when the visual feedback the waveform is providing isn't very useful. However, one of the most useful methods of selecting audio is the Tab key.

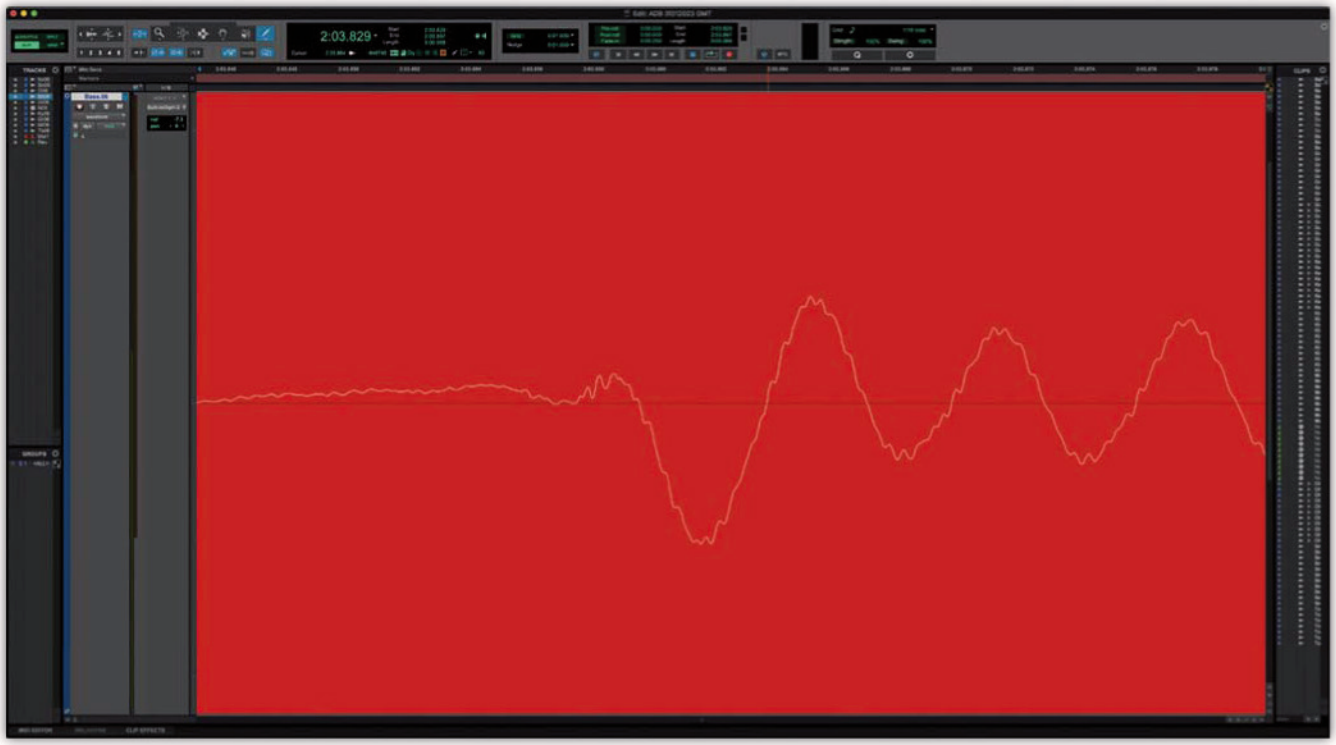
In its default state, Tab advances the Insertion Point (the flashing cursor line) to the next clip boundary. Use Option+Tab to go backwards, and even more usefully, use Shift+Tab to create or extend an edit selection. If you click the Tab to Transients button (bottom right in the Edit Tools section of the Toolbar), or use the shortcut Command+Option+Tab, you can change the behaviour so Tab will advance the insertion point (or, with Shift, the edit selection boundary) to the next transient in the audio waveform or to the next MIDI note. With the recent improvements introduced in Pro Tools 2023.3 you can Tab to an automation breakpoint too, a long-awaited improvement.

If you are trying to locate points in a project that doesn't use a click track or tempo map, and so isn't usefully editable in Grid mode, you can use Tab to Transients on a drum track to locate the points you need, then move your edit selection up and down to other, less transient-rich material, using P and ; (semicolon) in Command Focus mode.

Editing Tools

Most of the time, editing audio in Pro Tools involves three tools and, depending on who you ask, sometimes just one: the Smart Tool. The Smart Tool is polarising among Pro Tools users. Some claim they couldn't work without it, while others actively dislike it, but it is a distinctive feature of the program. With it, a user can flip between the three most-used tools — the Selector Tool, the Trim Tool and the Grab Tool — and create fades just by moving the tool to different areas of the clip (top half for Selector, bottom half for Grabber, either end for Trim, top corners for fade in/out and bottom corners for crossfades). It's quick and simple. You can further customise the Smart Tool by choosing one of the variants of the Grab and Trim Tools so, for example, you can have a TCE Trim Tool and the Object Grabber as part of your Smart Tool.

The big limitation of the Smart Tool, and a reason why I used to avoid it, is that you need to display your clips at a reasonably large size to be able to switch functions easily. Very small tracks don't work well. Something which



Zoom in on a clip far enough and the waveform changes to a single line. At this point the waveform can be directly edited using the Pencil Tool.

encouraged me to use it more is that it isn't an either/or choice; I use individual tools and swap to the Smart Tool much like I would change to any other tool. You can select it in the same way as you would select individual tools from the Function keys. To get the Smart Tool, hit more than one of F6, F7 or F8, in any combination.

Beat Detective

The ability to correct or adjust the timing for a performance is central to the editing process for music projects, and there are various equivalents to MIDI quantise that can be applied to audio content. Elastic Audio enables complex real-time time-stretching operations on audio content — anything from conforming loops to a session's tempo through to audio quantise operations and manual adjustments made using warp markers on Elastic Audio-enabled tracks. Pro Tools 2023.3 introduced a new algorithm in the form of zplane's elastique pro, which offers a huge improvement in fidelity compared to the standard real-time Elastic Audio algorithms and the rendered-only X-Form. All the major DAWs have an equivalent feature, but where possible, the best way to adjust timing is not to time-stretch at all, and Pro Tools has long had a powerful solution for this.

Beat Detective detects and separates the individual hits of a drum performance into clips in a semi-automated way and deals with the necessary smoothing of the resulting discontinuities between the hundreds of resulting edits as a batch process, meaning that the potential hours of manual editing are condensed into a few minutes of guidance. The drums won't suffer from the transient-mangling artefacts of time-stretching, because they haven't been time-stretched.

The Edit Window

The fact that if you zoom in far enough on an audio clip you will be able to edit directly on the timeline, even drawing in new waveforms with the pencil tool, is a revelation to users of other DAWs, which tend to have a separate sample editor. If you want to edit waveforms in full screen use Zoom Toggle (E in Command Focus mode) to really see what you are doing. For fixing simple clicks and pops, it's the fastest method you could wish for.

If you have a pure editing job on a single track of audio you might not even be using any plug-ins. If so, then there is no need to bounce the results out to a new file. Using the Edit menu's Consolidate Clip command (Option+Shift+3) renders the selected clips to a new whole file. If you just want that clip for use elsewhere, right-clicking on it in the clips list gives a contextual menu from which you can select Reveal In Finder to access the WAV file itself. If you need the file in another file format you can select Export Clips As Files and create an alternative version, for example an MP3 or an interleaved file at a different sample rate to the project.

I used to use a dedicated stereo editor in tandem with Pro Tools, but unless you need proper mastering facilities such as DDP, or features such as chapterisation for MP3s, there isn't much you can't do right from within Pro Tools.

Beat Detective provides an excellent alternative to Elastic Audio when conforming drums to the grid.

MATT HOUGHTON

Did you know you can embed parameter control knobs and some plug-in GUIs into Reaper's project and mixer tracks?

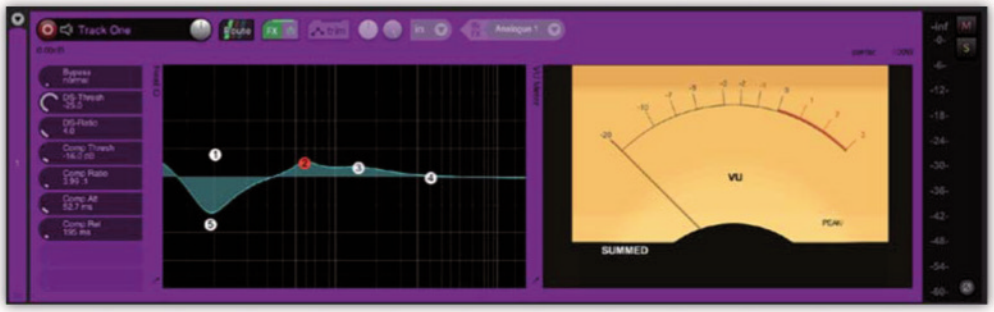
When it comes to customising Reaper's appearance, you can do much more than with any other DAW. Being honest, I'm not a big fan of the various 'themes' you can apply to the GUI — thoughtfully conceived and beautifully designed as some of them are, I'd rather we were all looking at broadly the same thing, since it makes explanation and communication between users so much easier! Still, even working within the confines of the default theme, there are lots of subtle but useful customisations that can make more efficient workflows possible.

One of my favourite features, which has been around for years, is the ability to embed control knobs and numeric value displays for individual plug-in parameters in the Track Control Panel (TCP). Say I have a channel strip, comprising instances of JS: De-esser, ReaEQ and ReaComp. Usually, that would mean I had to interact with three separate GUIs. But by placing the parameters I most frequently adjust in the TCP, I have instant acess to them without having to open/close any plug-in windows. More recently, in Reaper v6, another feature called Embedded UIs was introduced, that allows you to embed not just control knobs but also things like VU meters — the first screenshot shows an example of all of this.

In this workshop I'll explaining how to use both features, how you might combine them to useful effect, and how to overcome some of the small frustrations you'll inevitably encounter.

Track Controls

You can embed controls in the TCP for any parameter of any plug-in; you don't have to limit yourself to Reaper's stock plug-ins, though having said that some of them have particularly useful features for operation from the TCP.



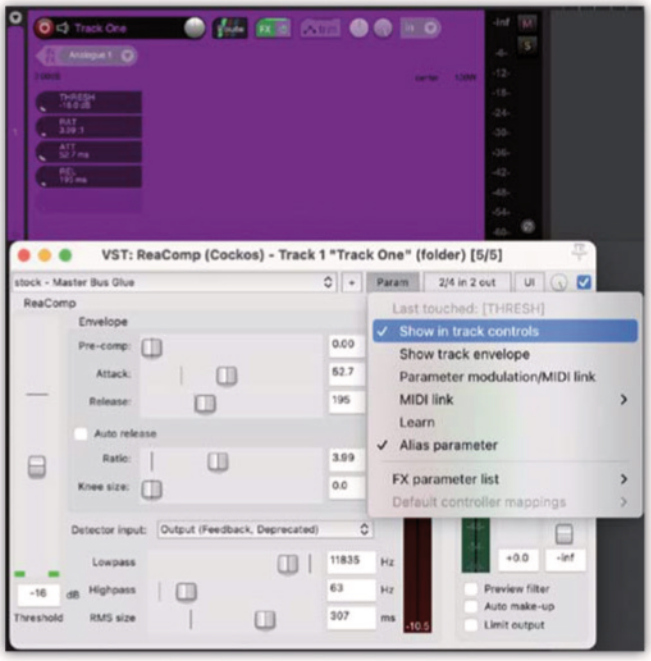
The finished channel strip control panel, with a combination of TCP controls and embedded plug-in GUIs.

We'll start by adding an instance of ReaComp. In its GUI, click and move the ratio control, then click on the Param button at the top of the GUI and select 'Show in track controls'. In the main project page, look in the control panel for that track — you should now see what looks like a rack of insert slots on the left, and in the top-most slot will be a knob for the ratio, along with a text label and the current value.

Click and drag on the ratio knob and you'll increase/decrease the ratio for that instance of ReaComp. It's also worth noting that if you double-click on the knob, the plug-in's GUI will toggle between open and closed, and that if you right-click you'll be presented with various options (such as parameter modulation) that are normally available from the GUI. Neat huh? So now let's rinse and repeat: add similar controls for the attack, release and threshold. The 'rack' will be populated with these knobs too, and they'll appear in the order the plug-in presents them, with the threshold at the top, bumping the ratio knob down a slot.

Things get more interesting when you start to add more plug-ins and to pick and choose the parameters you want to include in your channel strip. So let's add an instance of the JS: De-esser plug-in. We'll assume you want this to come before the compressor in your signal chain, so click and drag on the list in the FX dialogue window or the mixer's insert slots to change the order of your plug-ins. You can assign any controls from the plug-in to the TCP either as we did before, from the plug-in GUI and Param control, or, instead, by right-clicking on an empty TCP slot and selecting the desired plug-in parameter from the menu. For now, right click on a slot and select JS: De-esser / Frequency, then repeat the process to select the ratio. That should be enough control to do most things; if you find you want to do more detailed work you can always open the plug-in. Again, the sequence in the TCP reflects that of the signal chain, so even though you right-clicked in a slot after the compressor controls, the knobs for the de-esser will appear at the top.

Something you might notice is that we now have two separate threshold controls in the TCP with similar names. That's not helpful, is it? If you hover your cursor over a control, a tool-tip will pop up to tell you which plug-in it relates to, but that's still not ideal either. A better option is to rename the parameter, and you can do that by right-clicking on the one you wish to change and selecting Alias. You



Assigning plug-in parameters to the TCP controls from the plug-in GUI.

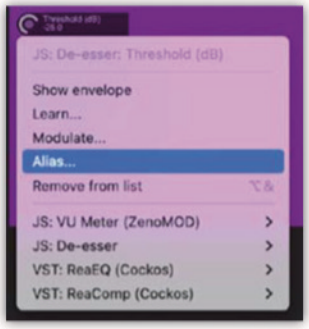
Once the parameter slots are in the TCP, you can add more controls from there too.

can rename as you wish, but it's best to keep names short to avoid the display being unhelpfully cluttered.

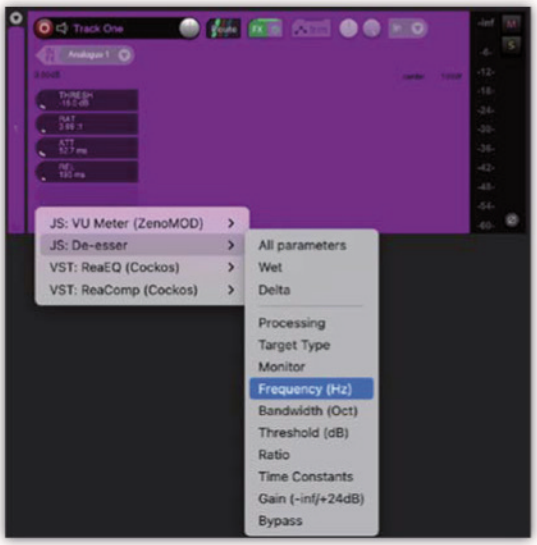
Finally, let's add a high-pass filter to our chain, courtesy of ReaEQ. Again, you can do whatever you want in the plug-in GUI but to keep things simple I like to have a dedicated one-band EQ for this purpose — I just change band 1's filter type to high-pass filter and delete the other bands. Then, rather than having a frequency control, I assign the plug-in's bypass control to the TCP. Sadly, you can't make the control a button (it's always a knob) but a quick drag toggles the high-pass filter on/off. You can't create an alias for the bypass control or assign each individual band's Enabled tick box to the TCP. But this filter sits at the top of our chain so it's not too confusing.

Embedded UIs

But before you go to the bother of figuring out how best to configure ReaEQ for TCP controls, let's look at the other feature I mentioend above: Embedded UIs. Introduced in Reaper v6, this facility makes it possible to include a compact, interactive version of a plug-in's GUI in the TCP (or, for that matter, Mixer Control Panel, though



Some plug-ins will have similar names for their parameters. Thankfully, except for the plug-ins' bypass controls, you can change that.



curiously not both simultaneously). Unlike the TCP controls, Embedded GUIs don't exist for every plug-in. In fact, there are plenty of Reaper's bundled effects that don't (yet) have an embeddable GUI. Also, the Embedded GUI doesn't precisely reflect the plug-in's main one — these are cut-down GUIs designed specifically for this purpose (which is one reason there aren't that many around yet... it would be great to see more developers support the format!).

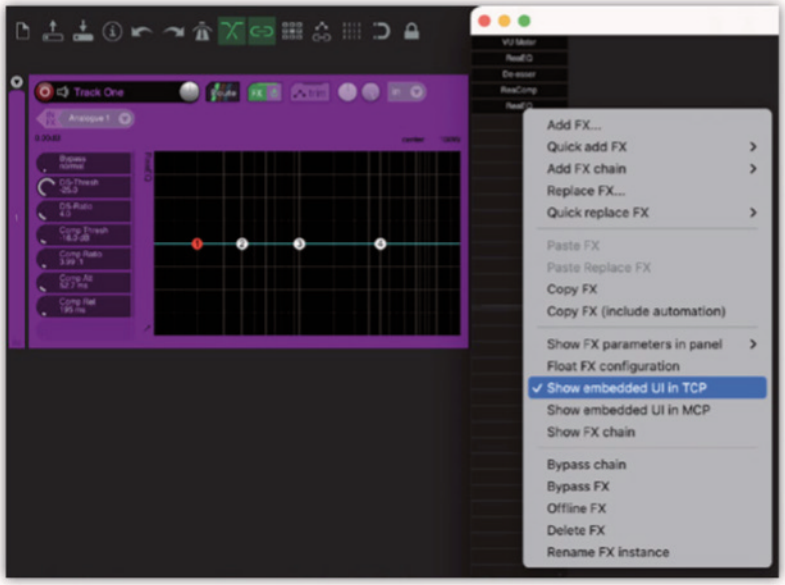
While your options may be limited, two that do already support the feature and are really useful for assembling our channel strip control panel are ReaEQ and JS: VU Meter (ZenoMOD).

Let's start by adding a tone-shaping EQ: place another instance of ReaEQ at the end of your signal chain, then right-click on the plug-in in your FX window list or mixer inserts and select 'Show embedded UI in TCP'.

Assuming your track is big enough in the project page (if not just drag to enlarge it) you should see a node-and-curve interface for the EQ. As usual, you can click-drag to change the frequency/gain, and modifier keys alter the behaviour: Shift allows you to change only the Q; Option/Alt allows you to change the frequency while locking the gain; and Command (on Mac — I assume it's Ctrl on Windows) locks the frequency so you change only the gain. You can also right-click a node to change more parameters, and a right-click anywhere brings up more options, for example the analyser slope.

This is quite a channel strip we now have in the project page, without needing to open any plug-ins, but one further addition I like is a VU meter. There's a perfectly good bar meter on the channel already, which can be configured to peak or averaging types, or even short-term or momentary loudness. But if I want to compare levels of tracks I sometimes find it easier to read several needle meters at a glance, and I also like that the VU meters are more sensitive around the zero point; you can instantly spot when something rises too high or drops too low.

The plug-in you need to do this is the aforementioned JS: VU Meter (ZenoMOD). You can embed it in the same way we did the EQ, and unlike the built-in channel meters, you can also drag/drop this to get a reading from any point in your insert chain. The only real downsides are that: if you have both the meter and the EQ embedded in the TCP, you'll need to make the track quite large to display both adequately (and this can be impractical on a laptop, for example); and that there are no configuration options available from the Embedded UI. Having said that, if you really wanted to go to town you could use Reaper's internal routing system to set up a weighting filter on the track's channels 3-4, to turn this humble VU meter into something more akin to a short-term loudness meter. But that's an idea we'll have to come back to another time!



Embedding a GUI for an instance of ReaEQ in the TCP.

DAVID RICARD

Logic's bundled delay plug-ins might be the only delays you'll ever need.

Logic's arsenal of delay plug-ins are a staple of my productions. And, while I own a basketful of third-party delays, I find I keep coming back to the dependable workhorses that I know so well.

In spite of the fact that I'm able to get reliably outstanding results with Logic's delays, I realised recently that I wasn't as familiar with all of their parameters as I thought I had been. With that in mind, let's reacquaint ourselves with the basics.

A delay is an echo effect where the delay time is the spacing between echoes, and feedback sets the number of times the echo repeats. And for the most part, that's how it appears to work. But to truly become adept at using this tool, a few distinctions need to be made.

The first of these is to understand that the effect is not creating an echo, *per se*. It's delaying the incoming signal before playing it. Think of it as buffering a signal, holding it for a spell, and then letting it run free. The reason we perceive this as an echo is because we usually hear the original source sound in addition to the delayed version. In fact, if you set up a pre-fader send from your track to a delay (100 percent wet, zero feedback), turn the source track's fader all the way down, and press play, all you would hear is your track... albeit starting a bit later. Essentially, it's as though you copied your track, slid the regions backwards and muted the original.

So what makes it sound like an echo? Well, firstly, when the delayed signal is combined with the dry source, we can now hear a call and response. To make the response sound more distant, we adjust its volume and tone.

And what about the number of repeats? Shouldn't there be a dial that says "Repeats 1-10?" That would make sense, but doesn't reflect what's going on behind the scenes. The delay only has one function: delay the incoming signal before playing it. That's where the Feedback control comes into play. Quite simply, it feeds the output back into the input. If you turn up the feedback to 100% in Logic's Stereo Delay, it will repeat the source track indefinitely in an endlessly annoying way. As you back the amount down, it sends less of the source back in, resulting in the classic dreamy effect we've all come to recognise. The feedback control is often underestimated, but with some very minor adjustments, you can dial in the exact amount of repeats you want.

Stereo Delay: the Global section (a) offers a number of creative routing options, the Feedback knob (b) sets how much of the output is fed back into the input, and the Deviation parameter (c) lets you offset the left and right timings against each other.

Stereo Delay

Now that we understand how the delay effect works, we can start to get the most out of it. The Stereo Delay plug-in is essentially two mono delays strapped together. The Stereo Link button forces any adjustments made in one chanel to be cascaded to the other. Don't think of this as a one-time setting: I usually start with Stereo Link engaged for broad settings, before disengaging it for fine-tuning. It's very common for me to set each delay with the same duration — either in milliseconds or note value — and then adjust the Deviation percentage up on one side and down on the other to create a bit of imbalance. And while the Stereo Link is handy for keeping the filters on each side consistent, having them at different settings adds some magic to the effect.

The Crossfeed knobs on each module introduces yet another dimension to the effect. Instead of adding a bit more of each delayed signal back into its own channel, the Crossfeed knobs let you add it to the other side. Once you start playing with crossfeed, you are practically entering the world of musical sound design, and the results can get quite trippy and hypnotic.

Now that you have a better grip on what these functions do, I hope you'll be less likely to reach for the presets. Not because they are substandard (they are excellent), but because this is one effect that it's worth the effort to roll your own to meet your project's needs. If that's too big an ask, meet me halfway. Set up the main parameters (delay times and filter settings) on your own and then cycle through the Routing presets from the pull-down menu. The second option, labelled Crossfeed, is particularly handy. It simply reverses the left/right inputs. So if you have a guitar that's hard-panned left, its delayed version will be panned right.

Tape Delay

When you consider all of the tape and tube/valve emulation plug-ins in today's market, you may wonder why Logic doesn't have any. But the surprising fact is that it does! However, rather than sporting graphics of spinning tape reels or glowing tubes/valves, Logic plays it cool with minimal fanfare.

The Tape Delay plug-in has always been my second choice for echo duties, but is starting to give the Stereo Delay

some competition. While it's presumably inspired by hardware units like the Roland Space Echo, Tape Delay ups the ante with some truly unique features. The plug-in forfeits some of the fancy routing options of its sibling and replaces it with a palette of vintage tape offerings. The difference in approach is that it focuses more on altering the sound of the delayed signal; the actual delay section is very similar to that of the Stereo Delay.

The Character section is where things start to get intriguing. The Clip Threshold is used to dial in some tape saturation, while the Tape Head Mode buttons give you options for simulating the tape head position. Remember, all of these parameters only affect the delayed signal, not the original.

The Modulation section gives you control over all of the highly sought-after inconsistencies of tape that used to give engineers ulcers decades ago. A hidden feature that I've fallen in love with is the Freeze button. When pressed, Tape Delay will repeat indefinitely. This means you can automate the Freeze button on the last word of a vocal and have it repeat as long as needed.

It's easy to get intimidated by parameters you don't truly understand, but you really can't mess up the LFO and Flutter controls. Just tweak until you like how it sounds. There's no real secret to this one.

Tape Delay works well when opened as a mono plug-in. It's perfect for vintage slapback echo. You can, of course, open it in dual mono and treat it like two units but, to me, that's not the best use for this one. When opened as a stereo plug-in you can take advantage of the Spread knob. This is another tasty automation opportunity to add drama and width.

I'm surprised that a spring reverb option isn't available, but that's an easy fix. More on that later.

The Character section of Tape Delay is where you dial in how much tape vibe you want, while the Modulation section gives you the ability to emulate the pitch inconsistencies that tape presents.

Echo

While Stereo Delay and Tape Delay are the thinking person's echoes, the simply named Echo plug-in is perfect for the don't-sweat-the-details types. It's essentially a mono (although it can be stereo) version of Stereo Delay without all of the routing options. Its simple interface is actually quite refreshing and it does the job without offering much in terms of tweakability. That said, the Color slider does offer a range of tones.

The humble and unassuming Echo plug-in is a good choice when you don't want to be overwhelmed with options.

It should be mentioned that with all of these plug-ins, as long as you open them on an aux track, you are free to add any processing you like to alter their sound. Distortion is a good option, as are filters and modulation effects like chorus and flanger.

Adding reverb to the delayed signal is a tried-and-tested manoeuvre to give a delay more space. Another great trick is to use the Stereo Spread plug-in after the delay to widen the effect. Additionally, don't feel limited to putting plug-ins after your delay; consider adding some before it. Besides the aforementioned ideas, try a pitch-correction plug-in before the delay for a vocal. How about the Enveloper to add some attack before your percussion hits the delay?

It's easy to forget or ignore how delays work, because they can deliver such usable results out of the box that we never need to examine what's going under the hood. But by getting more acquainted with the controls, options, and functionality, Logic's delay plug-ins become powerful creative tools.

DP is now fully equipped to take advantage of MPE. Are you?

MARTY CUTLER

Back in *SOS* October 2022, we learned some of the principles behind MIDI Polyphonic Expression, and how to set up Digital Performer (v11 and up) to record and play back MPE data. Since then, more MPE-ready synths as well as MPE-capable controllers have joined the party. So I'd like to share a few tips and impart a bit of advice for using MPE.

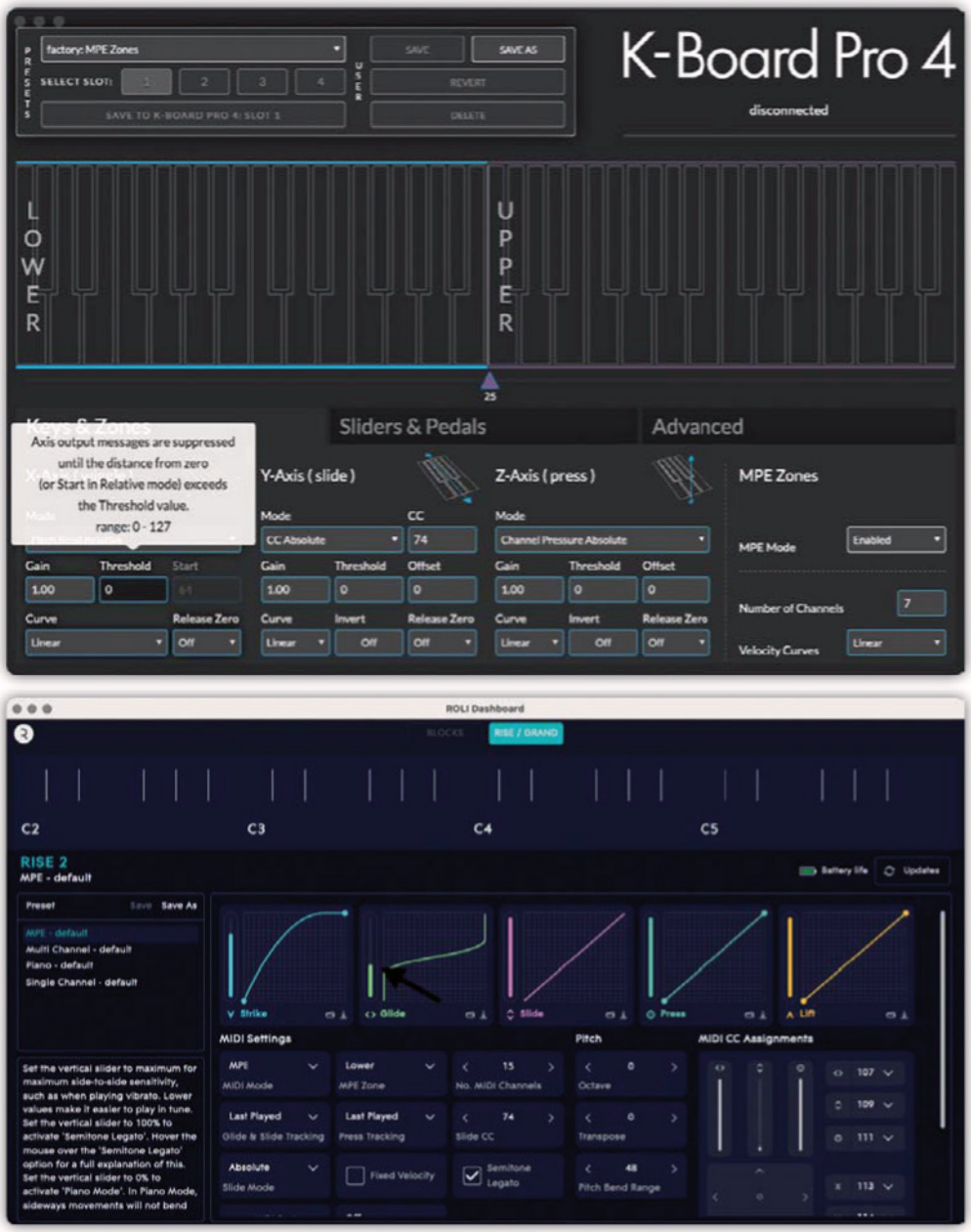
Getting Started

If you are approaching an MPE setup for the first time, you might need to make adjustments to your keyboard technique. Depending on your choice of controllers (I'm working with a ROLI Seaboard Rise 49 MkII and a Keith McMillen K-Board Pro 4), making music on an MPE controller can feel like tapdancing on a waterbed, both dynamically and pitch-wise. It's worth mentioning that ROLI's Seaboard MkII improves pitch accuracy considerably, but its response remains as accurate as your technique. Velocity and pitch-bend often elicit different responses on an MPE synthesizer compared with a conventional keyboard. You will probably need a lighter touch; fortunately, most MPE controllers can scale their velocity response or use custom response curves. There are other solutions to help your hands and MPE controller reach an agreeable compromise.

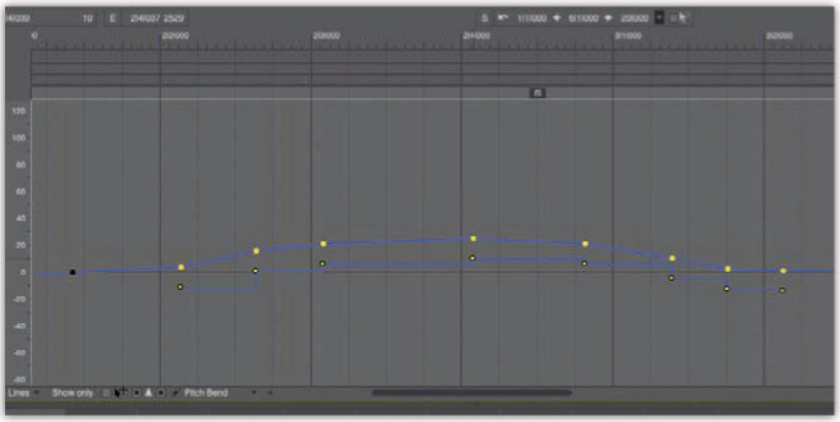
Relatively Speaking

You can be forgiven if you find playing an MPE controller a bit awkward at first; playing synthesizer keyboards is different from playing a piano, and playing an MPE controller requires yet another approach. Consider whether you need your synth to immediately respond to pitch-bend or other controllers upon contact with the keys. If you need your synth to sound in tune, you'll need to make sure that wherever you strike the key, pitch will be spot-on: either starting at zero, or some other pitch within the range you intend. The latter requires a precision initial touch on the key's horizontal axis, and can take some practice — violinists spend their lives making sure that their intonation is accurate. Likewise, you can alter the timbre of the synth gradually from the initial setting by sliding or moving your finger up the key, or you can land somewhere on the middle of the key to produce a radical change in timbre instantly. To address these issues, MPE provides Absolute and Relative modes for each gesture, including Key Pressure.

Glide (motion across the X axis of the key, which typically produces pitch-bend) provides a good illustration. In Absolute mode, landing dead-centre on the key will yield the expected pitch. Come down on either side of the centre and your note will be flat on the left, and sharp on the right. This is an excellent setup for creating a natural vibrato when you wiggle a finger from left to right, but difficult when you need to land on



Screen 1: MPE controllers such as the ROLI Seaboard range and Keith McMillen Instruments' K-Boards have software editors that can customise their response to suit your playing style.



Screen 2: MPE data can be displayed in DP's MIDI editing window as points, bars or lines.

Screen 3: The Change Continuous Data dialogue offers a number of options for manipulating your recorded MPE performance.

a specific pitch. Absolute mode assigns a range of values that always starts at zero at one pole of the axis, and increases or decreases as you advance your touch to the other pole. With Relative mode, any modulation along that axis starts at zero; pitch-bend starts at the initial pitch of the key, modulation starts at a neutral state, and so on.

Another way to refine MPE technique is by setting sensitivity thresholds. In this way, modulation won't start until you make more deliberate moves, and notes will not bend until you expressly move your fingers along that axis. Every MPE device I know offers software programming apps. The Dashboard app for ROLI MPE devices and Keith McMillen Industries' K-Board Pro 4 Editor offer easy-to-navigate graphical interfaces that simplifies the process, and a similar type of programming software (See Screen 1).

Bump Or Slide?

With all that said, sometimes you don't want gradual modulation. It might be preferable for sounds to instantly change timbre (for instance) at first touch. In that case, you'll want to opt for Absolute mode for the vertical axis. As with pitch-bend, you'll need to ensure that your MPE synth is set to respond in Absolute mode.

Another way to tame modulation is to dial back the synth's response. To control pitch-bend, for instance, try lowering the pitch-bend range, which defaults to 48 semitones with MPE. Conveniently, the McMillen K-board Pro 4's third horizontal slider bar adjusts the synth's pitch-bend range, and the fourth slider controls pitch-bend return time.

After The Fact

We can all get a bit overzealous in expressing ourselves in the heat of the musical moment, and sometimes we might find ourselves with an almost perfect take... almost. That can be especially painful when we have just one or two regions where we might have bent the notes just a little too far. Don't fret; Digital Performer has a few effective ways to polish up those near-perfect takes.

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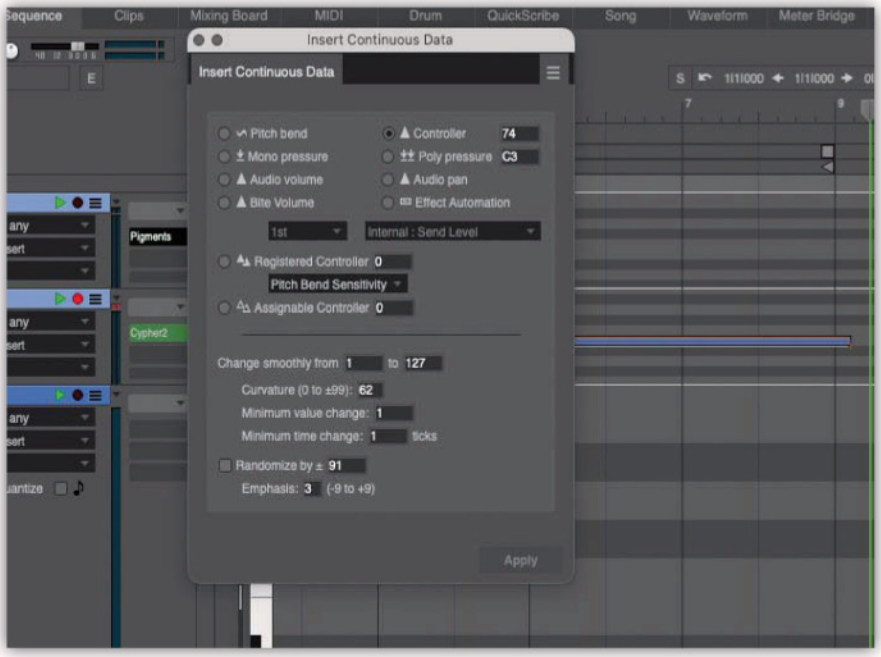
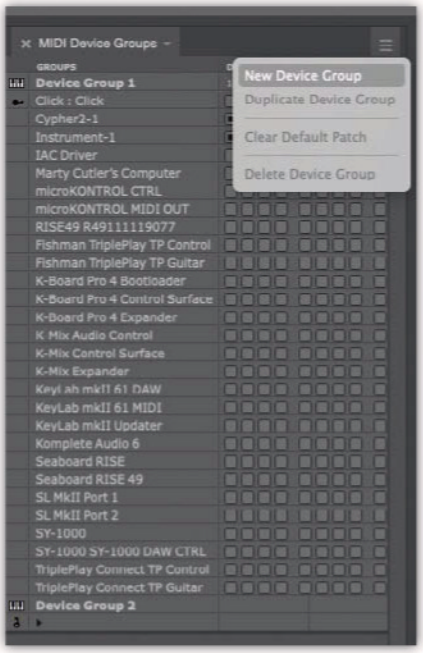


Screen 4: The Insert Continuous Data command lets you be precise about which gestures you apply MIDI Learn to.



Screen 5: Combine MPE with the MIDI Device Groups feature for a world of expressive fun.

» DP's MIDI editing window affords several handy ways to view your data: as points, bars, or lines. Each view has distinct advantages, but I prefer viewing continuous data as lines, as its simpler view shows the original selection while 'ghosting' the edit as it is performed (Screen 2). In the illustration I draw a marquee around the continuous data (in this case, pitch-bend), then click and drag to the proper level. It may take a bit of time to achieve precision, especially in the case of bending notes, but always remember that Undo is your best friend here, and a micro-amount of flatness or sharpness can add a touch of humanity to the mix. It's also important to select all of that group of events, rather than just the uppermost pitch-bend event; you probably want the passage to sound smooth and continuous, rather than having the pitch abruptly corrected. The same aesthetic holds true for other continuous controllers.

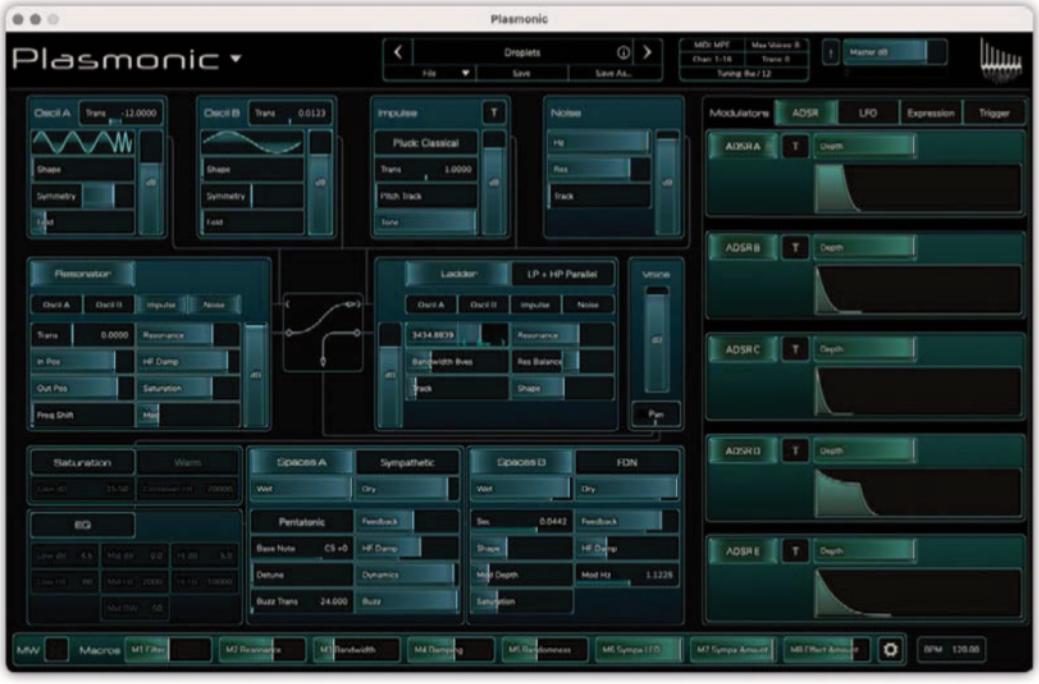
For more precision, select the data, go to Region / Change Continuous Data (in the DP menu), and choose the data type. If your otherwise perfect pitch-bend is sharp, select Limit to a Range and enter the highest and lowest values you desire. If it is flat, you can scale the data by a percentage or add or subtract a specific value to all events (Screen 3).

Let's Get Crazy

Now that you know your way around MPE instruments and controllers (and presumably Digital Performer), I'd like to share a few tips and tricks.

DP's region-editing features open a can of interesting creative options. For instance, record a staccato passage with your synth responding in Relative mode. Be sure to strike different positions on the vertical axis. Then try quantising just the controller data for some interesting rhythmic modulation possibilities.

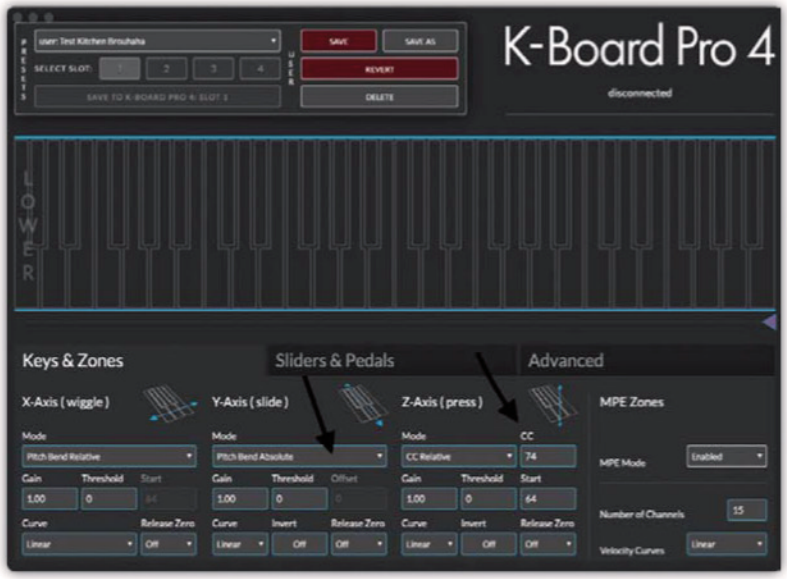
Some soft synths support MPE, but it may be that not all of



Screen 6: Stack a few instances of Rhizomatic Software's Plasmonic instrument for some MPE mayhem.

Just Ask The Axis

Here's an easy way to achieve quantised X-axis pitch-bend with the McMillen K-Board. In the K-Board Pro 4 Editor, it's a snap to reassign the Y axis to pitch-bend from its default assignment of MIDI CC 74. It's actually easier to manage polyphonic bending than using the X axis, and great for pedal-steel-like performances. In combination with Device Groups, you can achieve some true sonic mayhem. There's even a K-Board preset that offers both lateral and vertical bending, so you can get a nice combination of long, continuous slides and polyphonic vibrato effects simultaneously. Notice how I've also remapped polyphonic pressure to send CC 74.



the patches respond to all the usual gestures. For example, you might find a patch that would be perfect if only you could alter the timbre with Slide mode. Normally, your synth's MIDI Learn feature would be your go-to. Unfortunately, most MPE controllers send multiple gestures from your keystrokes, obscuring the message you need to send. Digital Performer's got you covered though, with the Insert Continuous Data command in the Region menu (Screen 4). Send a short stream of data to your synth and you are good to go. While we're at it, feel free to assign another parameter to respond to that controller: perhaps there's a wavetable index you can sweep, or unison tuning you can modulate. Not all parameters will respond polyphonically, but that's all right. You can create interesting timbral havoc with MPE Slide modulating filter cutoff and resonance simultaneously — it's all fair game.

To that end, DP's Device Groups can provide stunning sound-design possibilities. In short, Device Groups let you stack multiple synthesizers as a single playable MIDI instrument. Load at least two MPE instruments, go to Studio / MIDI Device Groups, and a window will pop up. Click on the radio buttons naming the devices you have loaded (Screen 5). Audition sounds to find the ones that you'd like to layer; your MIDI Device Group will reflect your choices. As each patch will have a different response to Slide and Aftertouch manoeuvres, the results can be startling. While you're at it, try setting the polyphonic pitch-bend range to a different setting so that the final result yields an interval from a single note. More synths with different ranges can yield chords! One of the beauties of synth plug-ins is the ability to load multiple instances, even of the same synth. I have had amazing results with Rhizomatic Software's Plasmonic (Screen 6); it's an MPE playground with no two patches sounding the same.

There's plenty more to explore. DP11, with its combination of traditional MIDI and MPE manipulation, has opened huge wellsprings of musical creativity. It's a powerhouse of expression, and I'm just getting started.

With this combination of Cubase's stock MIDI plug-ins, you can easily add variety and randomisation to your melodies.

They've been with us for years — but you really shouldn't overlook what Cubase's MIDI plug-ins have to offer!

JOHN WALDEN

Both the Pro and Artist versions of Cubase ship with a collection of 18 MIDI plug-ins, spanning creative, corrective and utility functions. They can be placed onto any MIDI or instrument track within the four slots provided by the Inspector's MIDI insert panel for the respective track. There are lots of utility-style functions available, but for this workshop I'm going to focus on a more creative task: adding variation to a melodic MIDI part, such as a lead or bass line. A number of individual plug-ins can serve a useful purpose here, but you can also combine them in some interesting ways. Particularly good candidates are the Step Designer, MIDI Modifiers and Density plug-ins. I'll explain how you can use these plug-ins below, and you can find some audio examples on the *SOS* website (https://sosm.ag/cubase-0623).

Start Me Up

Your melodic starting point might come from a part you've already recorded into a MIDI clip, but if you're stuck then Step Designer can be a great source of inspiration, especially if the piano/keyboard isn't your first instrument. In essence, Step Designer is a sort of streamlined MIDI grid editor, but it's designed specifically to make it easy to create short musical patterns. Once you've created a pattern, that pattern will loop on playback in sync with your project.

Step Designer's preset patterns provide a useful introduction to what's possible, but the key features include user control over the step size and count (up to 32 steps), the option to add swing to your pattern, the ability to add controller information for each step for velocity, gate (note length within a step), bank select and a single MIDI CC of your choice (CC10 for pan can be good fun). There are also useful tools for reversing patterns and moving them forwards/backwards by a step.

Wonderfully, each instance of Step Designer can hold up to 200 patterns, and you can easily copy/paste patterns. So if you create a good pattern as a starting point, you can duplicate it and then edit the duplicate to create a variation. Also, you can automate switching between the first 92 patterns by recording MIDI notes onto the Step Designer MIDI track. These notes themselves are not played by your sound source (only Step Designer's patterns will trigger sounds); they just serve to switch between Step Designer's patterns.

Creating a pattern is a simple matter of adding notes with your mouse, by clicking or dragging in the pattern grid. You can remove a note completely by simply clicking on it. Step Designer only allows you to place one note on each step, though, not chords.

Particularly effective (and speedy) is dragging your mouse across the grid editor to create a 'flow' of notes (perhaps a melodic line that sweeps up or down as it progresses). As we'll see in a moment, while Step Designer is not key/scale sensitive (so some additional tweaking of your 'note flows' might be

As well as creating note patterns, Step Designer lets you add controller data on a per-step basis.

required), our next MIDI plug-in can provide an automatic solution...

Fix It In The Mix

The 'correction' of any duff notes in your quickly created 'mouse drag' note patterns can be automated using the MIDI Modifiers plug-in. Placed in the MIDI insert slot below Step Designer, it can also do some other useful things to add both variety and a touch of 'human' to your Step Designer patterns.

The MIDI Modifiers plug-in has a number of different parameter sections that allow you to (surprise!) modify the MIDI data as it passes through, and two sections are of particular interest to us. The Random section allows you to apply random changes to any two of the pitch, position, velocity or length parameters for MIDI notes. For this example, I've used it to randomise the pitch of the incoming note upwards by up to 5 semitones (it also allows negative shifts, but I've not done that here). I've then also randomised the MIDI velocity of the incoming notes by ±10 steps.

With both these options, the settings you choose will influence the degree of variation that gets applied to your Step Designer pattern, with smaller min/max values producing more subtle changes (and *vice versa*). In terms of pitch, depending upon the major/minor emphasis of the overall piece, favouring positive changes only can often produce a more 'up' feel, while negative changes might create a somewhat darker mood.

To fix duff notes visit the bottom of the MIDI Modifiers panel. Here, you can specify a scale type and a root note (in this case, I'm using the Melodic Minor scale in E). Any MIDI note that arrives at this point in the processing chain that doesn't already fit the selected key/scale is automatically shifted to the nearest 'correct' note. Therefore, whatever notes you've defined within your Step Designer pattern, by the time they leave the MIDI Modifiers plug-in, they will at least be in key.

Incidentally, if you're starting a new project with this 'melodic variation' experiment and don't yet have a set of chords or key/scale combination in mind, it can be quite interesting to explore different scale/key settings to see quite what they do to your Step Designer pattern; some may sound terrible, but occasionally you'll be pleasantly surprised by what pops out. The Scale setting offers a huge range of scale types to choose from, each with its own musical flavour, and you can also choose to define your own scale should you wish; it's powerful stuff.

The Immensity Of Density

Having added some pitch and velocity variation to our Step Designer melody, the final option I'll suggest is the Density plug-in. This is super-simple to use and, popped into the MIDI insert chain after MIDI Modifiers, it provides a single 'note density' setting that allows you to either reduce or increase the number of notes played.

In this example, I've used a Density setting of 80 percent. This simply indicates that, for each note reaching the plug-in, there is an 80 percent chance that it will be passed onwards (so a 20 percent chance that the note will be discarded and won't play). This is done in a random fashion for each incoming note and can therefore be used to thin out your melody in a random fashion to a greater or lesser extent. If you automate Density's bypass button, you can use the effect to flip between your full melodic phrase (all notes present) and a sparser version of the same thing (a percentage of the notes randomly removed).

This overall MIDI plug-in chain can create some really cool results, especially with a plucky synth lead or piano sound that's providing a topline over a suitable chord sequence. Among the audio examples available on the *SOS* website you can hear the full MIDI plug-in chain in action.

If you pop open the Scales option in the MIDI Modifiers plug-in, you get access to a huge collection of scale presets as well as the option to create your own custom scales.

One More Thing

Well, two things actually! First, in the text above, I've described the process as applied to a lead melodic line. In that context, you can perhaps get away with some more extreme randomisation settings. However, apply more subtle settings (maybe just ±2 semitones of pitch randomisation and an 80-90 percent Density) and it can add some suitably modest variety to a Step Designer-driven bass line.

The second thing is perhaps the one frustration with the workflow described here: it's not straightforward to record the MIDI output resulting from the MIDI insert processing chain. Yes, the manipulated MIDI data will reach your virtual instrument for playback, but there's no simple way to capture the wonders of the randomisation as actual MIDI data so that you can edit it further or lock in some of the best results so that they play back in an identical fashion every time. There are ways and means, though, so this is definitely a topic I'll come back to explore another time!

Dying To Work In Music

Staying Healthy In The Studio Part 3: Nutrition

If music be the food of love, shouldn't we eat better while we're making it?

JACK RUSTON

"Eating well and keeping fit are absolutely essential to sustaining a career long-term. I put a lot of effort into both. None of us work better when our bodies are failing us": Tim Bazell.

In this series of articles, we're looking at the worrying decline in our baseline level of health, and how our lives in the studio can cause harm. In the first two parts, we discussed the related issues of sleep and stress. This month we're moving on to nutrition.

Nutrition is a hugely significant piece in the jigsaw of metabolic health, and it's been the focus of my attention for most of the past few years. Nevertheless, I raise it with regret and trepidation. If ever a subject deserved to join politics and religion in the list of those guaranteed to plunge even the most genial of friends into an argument that would make the Hundred Years War seem like a slight misunderstanding, it would be this one. The vast majority of the messaging we get around nutrition is sensationalised by the media and comes directly, or indirectly, from the food industry. We can be more confident about what isn't true than what is, and many of the questions we would like to ask are ethically impossible to answer.

Rather than trotting out endless pages of biological explanations, I'd like to offer this one fundamental piece of advice that saves us from an exhausting list of dos and don'ts: Eat nutrient-dense, whole foods.

Packing Them In

Let me briefly explain those two criteria, starting with nutrient density. Allowing a degree of simplification, we can think of our food as containing nutrients — protein, vitamins and minerals — and energy: fats and carbohydrates. The hormonal satiety signalling that prevents overeating and all its serious health consequences is skewed towards us getting enough nutrients, particularly protein. This makes sense, because not only is protein an essential building block of the body, but our traditional sources of protein are meat, fish, eggs and dairy, which offer the greatest density of micronutrients in their most bioavailable forms. If we ate enough protein, the chances are we would get everything else we needed in the process. Plant-based messaging from the food industry might try to imply otherwise, but it's not even close (although plants do provide some important micronutrients, and polyphenolic compounds). As the nutrient density of our

Just because the band have ordered takeaway food doesn't mean you have to.

If food has an ingredient list then, almost by definition, it's processed food.

meals drops, we begin to overeat energy in an attempt to meet our nutrient needs. This effect is termed 'protein leverage' and applies not only to humans, but widely across the animal kingdom.

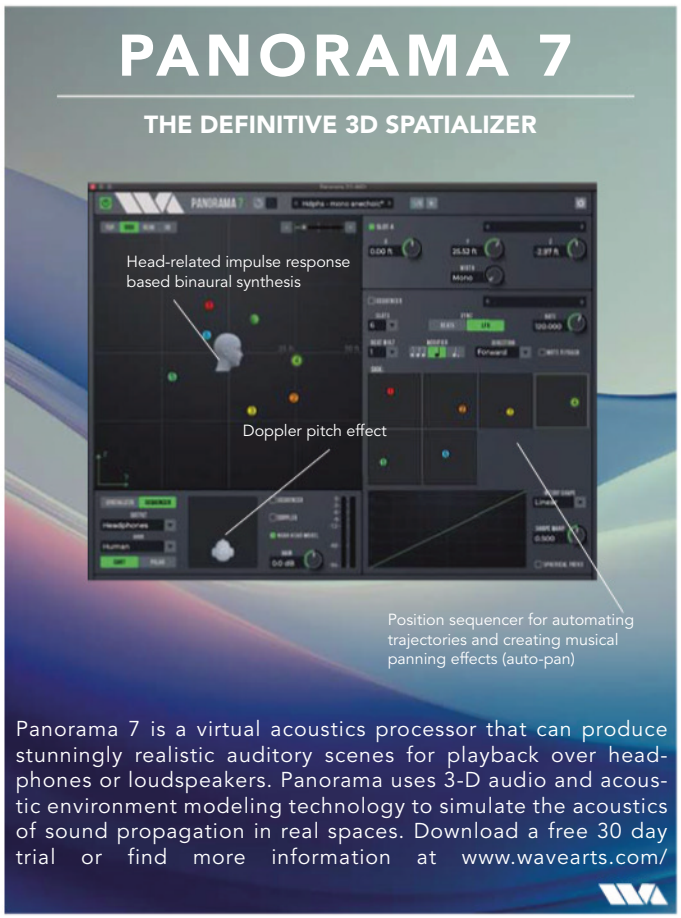
What about 'whole foods'? If a food doesn't have an ingredients list, it hasn't been deliberately formulated to make you eat more of it. Food companies engage in 'food engineering': the manipulation of certain flavours and textures to circumvent satiety. Products take advantage of the dopaminergic effect of sweet-tasting, high-energy foods, and our primal drive for sodium and the umami flavours that imply the presence of protein. The so-called 'Dorito effect' describes the way in which a food product can be made more compulsive by tasting like it contains protein, but at the same time less satisfying by actually containing very little. Food engineering is now a relatively mature science, and it's exceptionally effective.

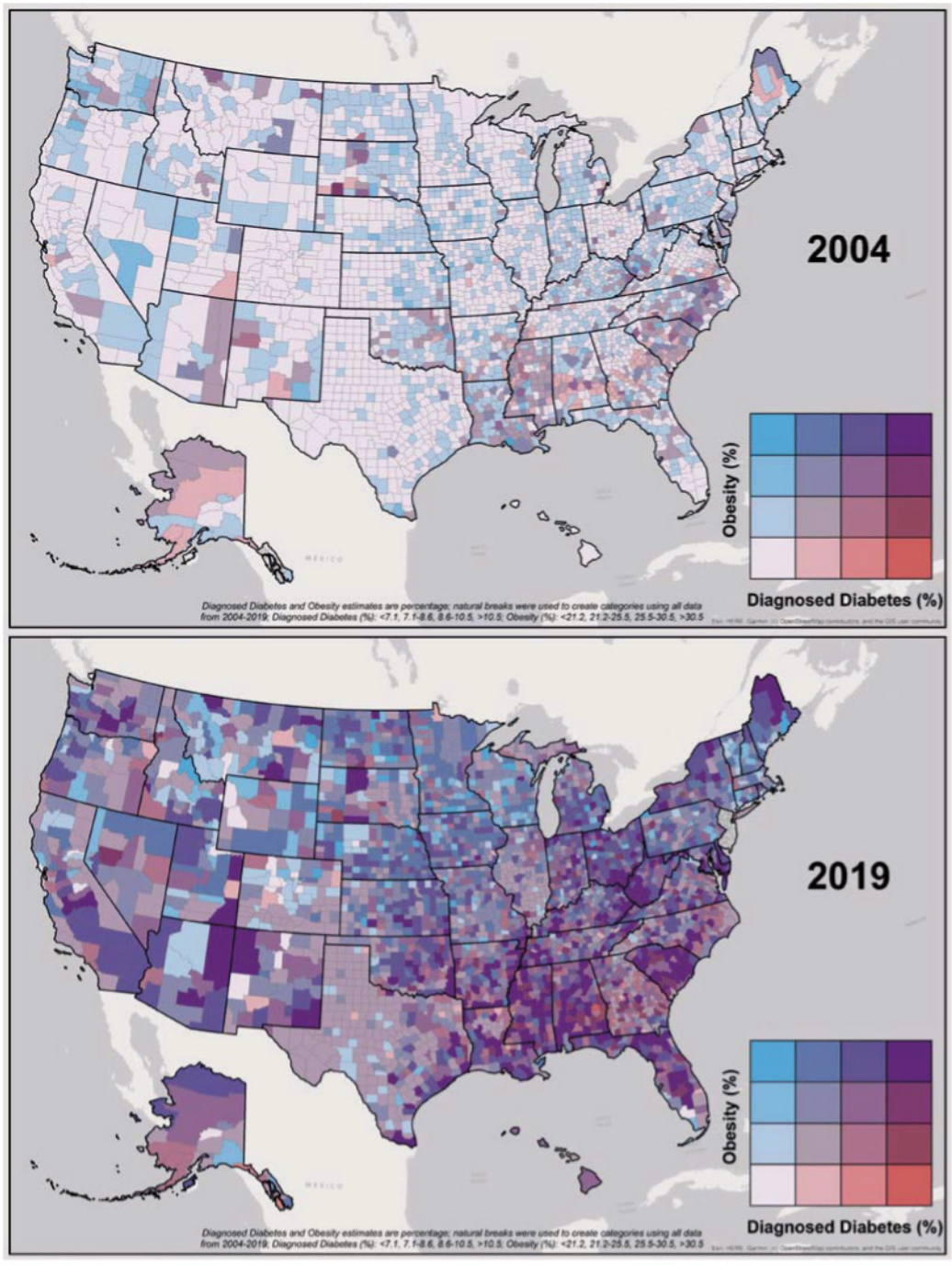
The Broad & Narrow Way

But confining our diet to only nutrient-dense whole foods is going to be difficult, isn't it? Yes, it is. Almost all ready-made restaurant and takeaway foods are highly processed and nutrient-poor. We will need to prepare most of our food ourselves if we want to have any confidence in its ingredients. For many of us, that's a tall order, but we mustn't let the perfect be the enemy of the good. Let's consider a continuum, with optimal at one end, and terrible at the other. What might those extremes look like? »

Studio Facilities

Many small studios lack a proper kitchen, and this is a significant barrier to eating well. If a studio has only a kettle and a microwave, the tendency is to gravitate towards packaged foods, ready meals and noodles. Even a lack of tableware and cutlery drives us towards foods that can be eaten out of their packaging, which is rarely a good thing in this context. It may be that we have to prepare food at home, taking it in with us on a daily basis. But the more we can provide in terms of a kitchen setup, the more flexibility that allows us. When it comes to designing our workspaces, it's worth bearing this in mind.





Shocking statistics: these official maps show how the prevalence of obesity and diabetes has grown in the USA over just 15 years, from 2004 (top) to 2019 (bottom).

» 'Terrible' consists entirely of food products and takeaways. It contains a lot of processed wheat flour, vegetable oil and sugar. It's low in quality animal protein, and relies on heavily processed poultry and pork, rather than fresh cuts of ruminant meat. In the 'terrible' diet, many calories come in liquid form, from high-fructose corn syrup in sodas, and sugar and fat added to beverages. The day starts with high-glycaemic carbohydrates that begin a cycle of blood sugar highs and lows, driving the need to snack every couple of hours or so, in order to prevent feelings of extreme hunger, shakiness or irritability. The feeding never really stops, as calories are always coming in from snacks or drinks. Meals are rushed, eaten on the go or while working. Digestion is poor, and symptoms like heartburn, gas or bloating, IBS and constipation are considered a normal part of life. Snacking continues late into the evening, causing disruptions in sleep and high cortisol levels overnight.

'Optimal' consists entirely of single-ingredient whole foods. Every meal contains a significant portion of high-quality protein that will provide nutrients, drive satiety and displace 'empty' calories. Carbohydrates come from fresh, seasonal, organic fruits and vegetables and perhaps some raw honey. High-glycaemic carbohydrate intake varies by season, and is significantly lower than in the 'terrible' diet. The optimal diet contains no processed seed oils, relying mainly on the fats that come naturally with animal protein, as well as cold-pressed, predominantly monounsaturated fruit oils like extra-virgin olive oil and avocado. Plant foods are prepared in such a way that maximises the bioavailability of nutrients, and minimises their natural toxins. Energy and focus remain stable regardless of meal timing. Most people feel quite satisfied by two meals a day, and there is rarely any need to snack. Meals are eaten slowly and mindfully in a calm state. Eating stops three or four hours before bedtime, and before the sun sets. Blood sugar is stable, cortisol is low overnight, rising again in the morning.

Small Steps

Most of us would look at the 'terrible' diet and think "I'm not that bad!", while the 'optimal' diet sounds difficult from a practical standpoint, and not all that much fun. The problem is that many of us are far closer to the 'terrible' end of the scale, and this is manifesting itself in rather frightening metabolic health statistics. We're all familiar with the ways in which

Make the time and effort to bring your own food to the studio, and you'll eat better and spend less money.

studio work can drive that sort of eating.

"An unhealthy lifestyle is absolutely intrinsic to the job," says Emre Ramazanoglu. "People are sedentary, living on takeaways and junk food because it's tasty and quick. I avoid processed foods as much as I possibly can. Not eating out is the biggest step — it has helped hugely."

We don't necessarily require an immediate seismic shift, but rather to see what small things we can do to move ourselves gradually along this continuum. Can we take home-cooked food to work? Can we cut out the soda? Can we replace cereal or toast with eggs? Can we get sushi delivered instead of pizza? Can we cut out the snacks? Can we stop eating a few hours before we go to sleep? Can we do this one day a week? Can we make it two? These little things add up to significant change.

When making dietary changes we need to be wary of diets. Our changes need to be sustainable and long-term. If we start thinking in terms of a period of 'dieting', we're just going to end up right back where we started once that period ends. Consistency wins the day. And if we cave and demolish a bag of crisps, we must be mindful not to throw the baby out with the bathwater: rather than giving up and telling ourselves that we've blown it and there's no point continuing, we can just accept that it's happened and move on with the best intentions. Suppressing survival-focused binge-eating instincts that have been developed over three million years when food is all around us isn't easy.

Do the best you can with what you've got. If you can afford it, and have ready access to grass-fed, pastured animal foods and organic fresh fruit and vegetables, that's perfect, but for many, these premium options are unrealistic. Just know that those single-ingredient whole foods are always going to be better than the packaged products.

The nutrition journey is a perilous one, and we must be wary of traps. These mostly come in the form of processed 'health' foods, and food products that fit a certain dietary approach 'on paper'. Avoid granola bars, protein bars, shakes, plant-based, dairy or gluten-free alternatives, nut or oat milks, fruit juice, sugar that's pretending to be something else — you get the picture. This is why the notion of whole, unprocessed foods is such a good one. It protects us not only from obvious junk, but also from junk marketed as 'healthy'.

It's important to note that on a vegan diet, it's not possible to meet your nutrient needs without careful supplementation. There's a reason that no human society has ever been vegan. The use of processed foods like combined plant proteins may also be necessary, especially given that the very low bioavailability of nutrients will require greater quantities overall. If you wish to be vegan, take professional advice on how to construct your diet. Simply eating plant-based foods and hoping for the best is likely to lead to problems in the medium term, much as any move away from the typical diet of junk food is going to be of benefit in the short term.

Next month, we'll finish this series with a look at movement.

Tips:

- **Eat at least 30g of quality protein in every meal or snack.**
- **Make your food at home and take it to work with you.**
- **Provide as extensive a kitchen setup as is practical.**
- **Don't drink your calories.**
- **Don't be tempted to eat every time you see a band member doing so.**
- **Be wary of packaged foods, and anything with ingredient lists.**
- **Don't let slip-ups derail your progress.**
- **Don't go 'on a diet', but rather change the way you eat for good.**

Avoid Smoothies

The trend of blending up vast piles of raw spinach, kale, nuts, juices and whatever other 'superfoods' are around can do more harm than good. We need to be a little bit respectful of the plants in our diet. They can be enormously beneficial, but they're not invariably safe in any quantity. They've successfully defended themselves against being eaten to extinction by virtue of containing toxic compounds. Some of these can even be good for us, but only when eaten in appropriately small quantities. Our ancestors went to great lengths to make plants safe to eat by cooking, peeling, fermenting, soaking and sprouting, and it seems that much of their wisdom has been forgotten along the way. Smoothies let us rapidly consume raw plants in quantities far greater than we could ever have obtained or eaten in their whole-food form. This can drive issues with digestion, intestinal permeability, kidney stones and thyroid issues, as well as dermatological problems. It's also worth noting that drinking our food negatively affects our satiety signalling: we don't feel as full.

TalkBack

Mikko Gordon

WILLIAM STOKES

Born in the UK, engineer and producer Mikko Gordon grew up in Helsinki before returning to study at Goldsmiths, University of London. Since then he has formed a close working relationship with Radiohead super-producer Nigel Godrich, who he first met through his wife, Laura Bettinson (aka Lau.ra), who is producer, DJ and vocalist in Godrich's own outfit Ultraísta. As well as touring with Thom Yorke and Phil Selway as live engineer for their respective solo projects, Mikko engineered the debut album by the Smile, the trio comprised of Yorke, Jonny Greenwood and drummer Tom Skinner. He also toured with the band and mixed their live albums.

"Originally, I was thinking, 'I want to be in a band,' but I realised very quickly that I hated the democracy of the process!" Mikko laughs. "Wearing the producer's hat and working in the studio is much more interesting to me."

At moment I can't stop listening to

I'm listening to a lot of electronic music at the moment; Luke Abbott is a current, huge favourite. 'Swansong' would be a song I'd recommend to check out. The sounds are just so... organically synthetic is the best way to describe it: you know, they're modular, but they're so real! The way they jump out, all the drum sounds and things. It's amazing.

There's also an Australian band I recently discovered called the Lazy Eyes, who are wonderful! Really young band, just doing really interesting stuff. And Nilüfer Yanya would be another one for me. She's an amazing UK artist, just her sound and her vibe. I've been listening to her a lot. I did a session with her and her band as part of another series of the *From The Basement* programme.

The project I'm most proud of

I'll have to say engineering the Smile album [*A Light For Attracting Attention*]. That was just an incredible experience for me. Thom Yorke and Jonny Greenwood are just such incredible artists and musicians, and Tom Skinner as well — such a phenomenal drummer. It was a really fun and challenging project at the same time. We recorded to tape, working quite quickly. It was a very, kind of, free environment for the band. Just to try and figure out stuff, to play things live together. I think, especially after Thom making a lot of electronic records, the band were just really enjoying having three people in a room playing music together. And the kind of excitement that came from that. We were trying to minimise the role of computers, you know, recording to tape and just keeping things moving. It was just an incredible album to be a part of, to be creating sonics around these songs that were coming together.

The first thing I look for in a studio

It's definitely the vibe and the feel of the space. A lot of studios feel quite clinical and sterile, and not like creative spaces. I'm happy to record in houses or any location — they don't even necessarily have to be studios. So how it's going to feel is important.

After that, it depends on the project. A big control room, the monitoring, you know. If we're talking about technical things, it's very important to be able to trust what you're hearing. And I'm always happy to bring in a set of speakers myself, if I need to. And then, on the equipment front, I'm old school: I love an API or Neve preamp, an 1176 compressor, Neumann microphones. But again, if the space is right then I can bring that stuff with me.

The person I would consider my mentor

Well, definitely Nigel Godrich. He's been an incredible mentor and friend. I've learned so much from him about how to think about music and how to approach things aesthetically. I'd also name Mandy Parnell, a mastering engineer who early on in my career was incredibly supportive, gave me a lot of her time. I would just go and attend sessions and, kind of, sit there listening to all the problems in my mixes being revealed!

She would talk about some of the things that I need to think about or look at, and then go ahead and fix them all, and make the tracks sound amazing. I would even come in sometimes to listen to a mix I was unsure of on her system, then go away and work on it a bit more. Or even tweak something there in the session. She was never in a rush to move to the next thing; she always had time to help me, which was amazing. Her support helped me to get better with my mixing.

Finally it would be Tchad Blake, who I'm also very grateful to for having given me his time over the years, listening to things I've been working on and letting me see him in action. That's been absolutely incredible.

My go-to reference track or album

I worry that I'm going to come across arrogant with my answer, because I tend to reference music that I've worked on myself! Because I know how it sounds inside out, and I know where I've wanted to go sonically with it. So usually, when I'm walking into new studio, I'm going to be listening to a few songs from projects I've worked on. But if I was playing something by someone else, there is a Four Tet remix of Ultraísta, which sounds phenomenal. It's got this beautiful, deep low end. A great-sounding track. And also I get to hear my wife singing while I'm doing it, so it's win-win!

I might be going on a tangent here, but it's interesting how there just isn't really an 'average frequency balance'. You listen to

a record and it's got no low end, but it still sounds great in this really... compact kind of way. And then you've got records with like, huge bass. Maybe all the highs are really muted. So everyone needs to figure out what their ideal is. You can have stuff that sounds amazing, but you're thinking, 'I would never make it sound like that.' But I can still really respect how that's been done.

My top tip for a successful session

Making records is an intense experience. And I think it's really important that artists feel like they're able to try out new ideas and to experiment — and to fail. It's really important to have that comfort factor, to be able to fail, because through that you get somewhere. You go to new places, and that kind of freedom brings out creativity. So I think it's about creating the kind of right space and environment for that. That's one reason why I'm never keen on people bringing in film crews to recording sessions, because it changes everything. Suddenly everyone's considering what they're saying, and it's kind of counter-productive to the creative process.

Beyond that, sonically, I want the record to start taking shape as early as possible. So I'm a big believer in committing to decisions and sounds early on. I want people to be going in to play and putting their headphones on, and to have it feel like a record is already there.

I get stuff to mix sometimes and there are, like, 30 microphones on the drum kit, but there's nothing defined. And it just sounds like a boring drum kit! And it's like, we're just pushing the creative decision process way down the line. Whereas if that decision had been made earlier on, then the other creative decisions would have started building on top of that, and the whole end result might have been more interesting.

The studio session I wish I'd witnessed

Well, growing up, the three big bands that I listened to a lot would have been Radiohead, Arcade Fire and the Cure. And I got to work with Thom and Jonny already. I've made a record with Arcade Fire already. So the Cure is definitely the last one! I'd love to make a record with them. But when it comes to sitting in, it would be on the sessions for the album *Disintegration*. That would have been an amazing experience. Just the atmosphere and the feel. That record is very dark at times. And I'm curious about whether the mood was really sombre, or whether they were, you know, just laughing and joking around in those sessions... just what the feel was, what the goal and drive for everyone aesthetically making that record was.

The producer I'd most like to work with

I mean... I've already worked with him! I'm very fortunate to have worked with Nigel Godrich a lot. But Tchad Blake would be someone else. And Dave Fridmann. He's worked on loads of great records that sound really interesting and creative, the way he brings that interesting kind of psychedelic element.

The part of music creation I enjoy the most

It's just the act of working together with people to create something. You know, creating music that will hopefully resonate with them and have a connection with them. It's great when you're coming up with sounds and you can't really tell what something was or where it came from. And hopefully that will leave listeners wondering how those sounds were made.

The advice I'd give myself of 10 years ago

I feel like, 10 years ago, I was only just getting to a point where I was figuring out knowing what I'm doing. I think my advice would be to be fearless. Keep experimenting, and just stick at it.

INSIDE TRACK

SECRETS OF THE MIX ENGINEERS

Marta Salogni

Analogue tape played a key role in the production of Depeche Mode's album *Memento Mori* — even though it was recorded to Pro Tools.

PAUL TINGEN

Marta Salogni's unique approach to music production has seen her recognised three times by the Music Producers Guild, with the 2018 Breakthrough Engineer of the Year Award, the 2020 Breakthrough Producer of the Year Award, and last year with the UK Music Producer of the Year Award. As well as a wide range of leftfield electronic, alt-pop, art and punk rock acts, Salogni has made her mark on the mainstream with credits like Björk, Bon Iver, Black Midi, Sampha and Animal Collective. Prior to going it alone, she also worked with David Wrench on mixes for Frank Ocean, the xx, Goldfrapp, FKA Twigs and David Byrne.

This Spring, Salogni achieved her most significant mainstream success to date when Depeche Mode's 15th studio album, *Memento Mori*, went to the top of the charts in a dozen countries. Salogni recorded the entire album, and mixed it at her Studio Zona in East London.

Zoning In

The studio is a hybrid of the traditional and modern, with a prototype 22-channel Studer console from 1974. Its most striking feature is Salogni's collection of a whopping 12 quarter-inch tape recorders (see box), the result of a slightly unorthodox career trajectory. Growing up in Italy, close to Milan, with a keen interest in art and music, she moved to London in 2010 to attend a course at the Alchemea Music Production College. Around that time she became interested in synths and tape machines as ways of expressing herself musically.

"I love the fact that on a synth you can create sounds that didn't exist before. I love traditional instruments as well, but it takes a lot of time and patience to learn how to play one. Whereas, with synths you can switch them on and immediately experiment

'Ghost Again'
Written by Martin Gore & Richard Butler
Produced by James Ford

with sound in a way that's instantly personal. Plus they are intuitive and creative. It's the same with tape recorders. I became fascinated by them, and all the different things you can do with them, even though they're originally built to record."

After Alchemea, Salogni spent several years in several different studios in London, including at Dean Street, RAK, Strongroom and Mute, and assisted producer Danton Supple and mixer David Wrench. For a while she had a studio at Mute, but since the beginning of 2018 she's been working at her first completely independent facility, Studio Zona. While most of her work now consists of mixing and/or producing, Salogni also uses tape recorders on the live stage, performing alongside other musicians or alone.

In The Studio

The roots of her involvement with Depeche Mode go back to 2015, when Salogni assisted Danton Supple on the album *Angels & Ghosts*, by Soulsaver and Depeche Mode singer Dave Gahan. In 2020 she mixed the follow-up album *Imposter*. In recent years, Salogni has also mixed several projects for producer James Ford, known for his work with Arctic Monkeys and Florence + the Machine.

"James called me in the Spring of last year, and asked me, 'I know you do mostly production and mixing these days, but would you be interested in recording Depeche Mode with me for a new album?' I love Depeche Mode, so of course I said yes. But then, Andy Fletcher [*the band's keyboard player*] passed away, and all we could do was wait for updates. In the end the decision was made to move forward with the album, also in honour of Andy.

"So in July of last year I travelled to Martin Gore's Electric Ladyboy studio in Santa Barbara. Martin's studio is exceptional. It has everything you can possibly need, all the synths that you can imagine. It's like a playground and a place in which I felt very inspired. It's also really well laid out from a technical point of view, with a U-shaped, custom-made rack of gear, with preamps, dynamics and distortion on one side, effects on the other side, and then there are all the Euroracks, old Moogs, and the Roland System-500, and so on.

"There are Barefoot monitors, and the desk is an SSL Nucleus, so not analogue, but we had enough analogue gear to make up for this. All the polyphonic synths are in a second room, where we built an improvised vocal booth. In the first half of the sessions it consisted of some mic stands and rugs that we found, and in the second half Martin got someone in who built a proper heavy curtain booth. There's also another room with more synths and keyboards, and pedals, guitars, basses and amps. We had everything we needed!"

Super Sessions

The core team at Electric Ladyboy consisted of Gore, Gahan, Ford and Salogni. "We were really tuned in," she says, "and had a beautiful time together. We experimented a lot. They were really open to experimentation, and excited whenever I suggested something new. On one of our days off I said, 'I'm really missing my tapes machines because they're fundamental for my creative process. Do you mind if I find some?' They were up for that, so I made several phone calls, and I found some at Wiggle World, a brilliant studio in Los Angeles.

"The studio kindly let me borrow three tape machines: an Otari MX5050, an Akai and a Sony, all two-track quarter-inch. I felt »

Tape To Tape To Tape

"I've got five Revox tape recorders," explains Marta Salogni, "one PR99, two B77 and two A77s. I also have four Akai 4000DS MkII tape recorders, two of which I use for spares and experiments, an Akai 1721, a Ferrograph 5A, and a TEAC four-track. I've also got two Watkins Copicats, and a weird Davoli Echo Mixer. Davoli is an Italian company that used to make Binson Echorec-style stuff.

"I always try to make gear do stuff it was not designed to do. I like to use my tape recorders as effects, for example to add saturation and distortion. Or they can act as delay units, or to add textural feedback, or to create loops and polyrhythms, or I can use them as pitch-shifters thanks to their varispeed. I feel I can do pretty much everything with them, really. There is a lot of freedom within that process. They're very multifaceted and versatile.

"My Revox PR99 is the top-of-the-range version, with XLR in and out, and I may send a full mix out to it to add some analogue quality, or use it for tuning specific delays, since it has a very precise varispeed function. The B77s have RCA and jacks for input and output, and I may use them to create loops, cumulative loops, and delays. Tape will travel from one B77 to another, with one set to record and the other to playback. It can feel like a maze or labyrinth in here sometimes!

"I do the same with the Akais, but obviously they have a different sound, and also, the tape tension is different. Sometimes when I have tape going from one Akai to another, the tape will lose tension and this creates modulation that will then be regenerated, and that contributes to the overall effect.

"I used the two-tape-recorders effect on many tracks on the new Depeche Mode album, including the opener 'My Cosmos Is Mine'. David Gahan's vocals came back from the second tape machine, and when I then feed it back in, it will start interacting with what was there before, creating a cool effect. I like that process during the compositional and production part of a record, because it can inspire new things.

"Working with the tape recorders can bring unexpected results, because things are never the same twice. I often use tape recorders live when recording, because a player or singer will feel like they are responding to something outside their timing, and duets with the machines start to happen. It keeps things refreshing and interesting. I also use tape recorders to create harmonies and layers. It works great on strings, for example, when they play long notes, and these go from one tape recorder into another, and the delays create harmonies and layers.

"Also, it's nice to get out of the chair to work with the tape recorders. I enjoy being able to physically work with machines. There are projects in which I use a lot of tape, which is RMG SM911, but also projects on which I don't use tape at all. It depends what people want. People do tend to come to me for creative input, and because of my way of working, and that means that I tend to have a lot of freedom. I can do pop, which works more according to a recipe, but also because I tend to work with more leftfield acts, I can bring a different angle to the traditional approach. It's what makes me wake up every morning and feel honoured and glad that I'm alive, because every day is different!"

A few of Marta Salogni's 12 reel-to-reel tape machines.

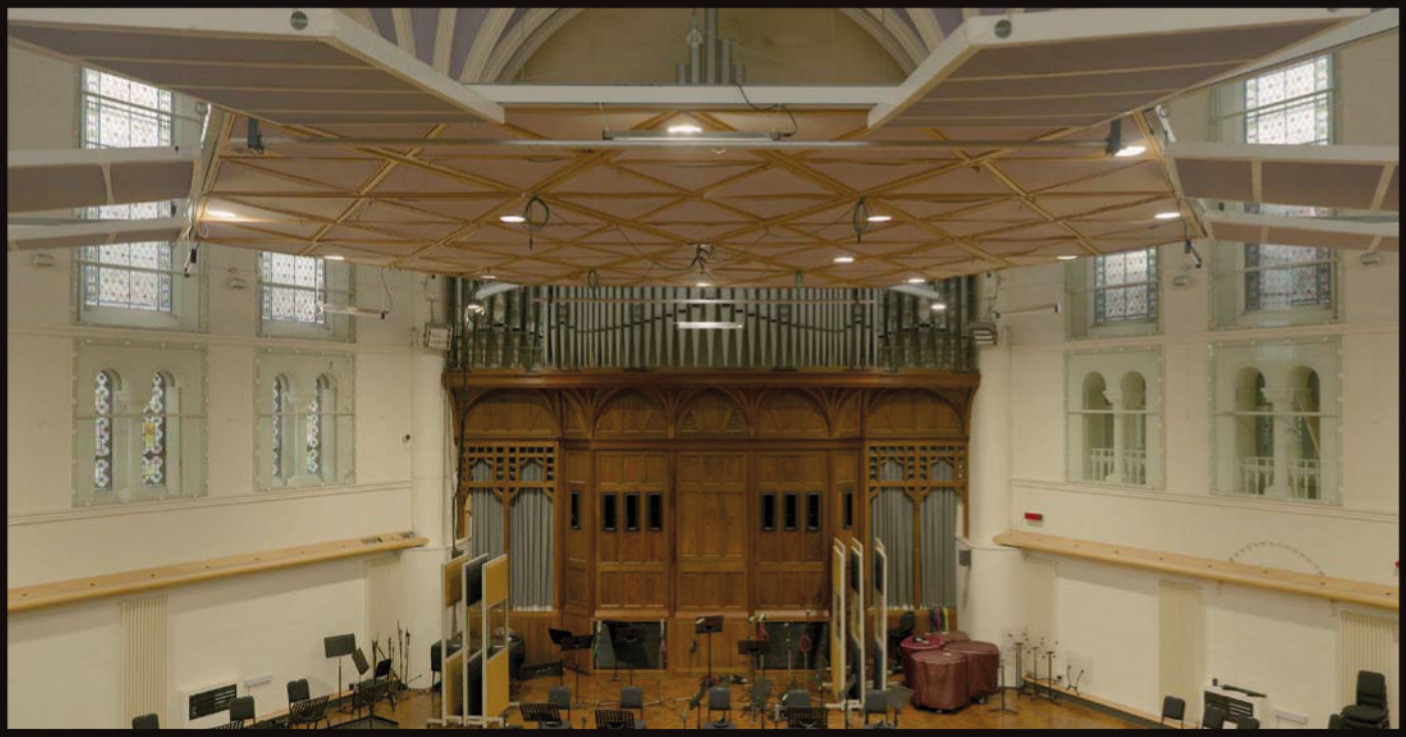
» like my best friends were back! I started experimenting with them and making loops and realised that there was a lot of scope to use them on this record. The others also really liked it, and so when we returned for the second set of sessions, which began in September, Martin had bought two Revox A77 tape machines for his studio."

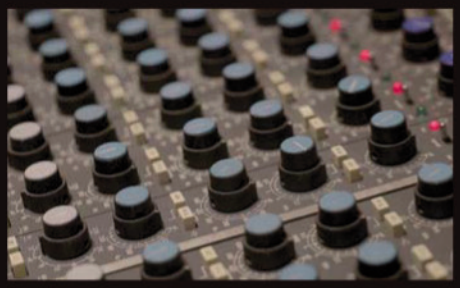
The tape machines were used alongside many other sources of effects. "I recorded Dave's vocals with a Neumann U87, which went into an API preamp, an LA-2A, and finally a Massive Passive EQ. On top, Dave loves to sing with effects already set. So before he came into the studio I set up a few chains of effects, some of them quite extreme, and he loved that. He was like, 'I can project into these spaces.'

"A lot of these spaces came from the tape recorders, but in addition I also used effects units like the Eventide H3000, the Eventide Eclipse, the Mu-Tron Bi-Phase phase shifter, the Publison IM90 Infernal Machine sampler, an Overstayer MAS, a Roland Dimension-D, the Ursa Major Space Station, and AMS RMX16 and Lexicon digital reverbs. We used the Dangerous Liaison to line some of them up in a chain, and this also allowed us to mix them up. I'd then record his vocals dry in Pro Tools, and printed all the effects separately.

"We created similar chains, again also using tape recorders, for the synths, drums, guitars, and also the strings. There are no dry strings in the record. We went to Shangri-La [*Rick Rubin's studio*] to record the strings with Davide Rossi, as well as some drums played by James, and more vocals. We brought a lot of gear, including tape machines, to add effects, and I used the bathrooms in Shangri-La as echo chambers on the strings. In general, we recorded everything through a lot of effects, because we wanted to make it sound interesting and different.

"I also was honoured to receive a writing credit on one song, 'Speak To Me'. It was »





SOUND ON SOUND

VIDEO

VIDEO DOCUMENTARY ORIGINALS

IN ASSOCIATION WITH

UNIVERSAL AUDIO

THE AIR FAMILY

Thirty years ago, visionary producer Sir George Martin saw the need for a London studio that could record cinematic film scores. AIR Lyndhurst remains a favourite destination for Hollywood composers, rock bands and pop stars, and it's just emerged from a stunning renovation. We visited the converted mission school in Hampstead to tour the facilities and meet the staff who keep AIR at the top of its game.

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» written by Dave, and James and I literally locked ourselves in the studio one day and completely remade his demo, and presented it to him. He loved it, so much that he gave us a writing credit on it. We changed key, changed tempo, and took out all the instruments, and built the arrangement back up from the vocals. James and I used the EMS Synthi A for a lot of this, for example creating the high-pitched drones and the heavy kicks at the end."

Console Capers

After two and a half months of recording sessions at Electric Ladyboy, Salogni returned to her Zona studio for the final mix of the album. "I was really looking forward to working in my room with my desk, speakers and sub, which I know well, and to make sure that all the elements in each song could co-exist and gel together, not only track by track, but also as an entire album.

"As I was mixing, I sent off my first draft to everyone, and after getting their feedback, I imported that mix into the mix session for the next track, so I could A/B between the two. I did this for the entire album. We wanted a consistent sound across the album, unless it was a creative choice for something not to be consistent."

The gear at Zona that Salogni uses to mix is kept deliberately simple — other than the 12 tape recorders, of course. "I wanted to keep the studio quite minimal, so I looked for a desk that had everything that I needed. My 1974 Studer desk is actually two Studers put together, united by a meter bridge. It has 22 channels, eight groups, and has been modified to have direct outs, so it integrates well with my Avid HD I/O for Pro Tools. I can mix either fully analogue, hybrid, or fully in the box.

"The desk has EQ on every channel, and limiters, so I don't need much other outboard, other than the tape machines. If I need anything specific for a session I just hire it in. I've got some things on my wish list, but for now I'm happy. I know everything in this room inside out, and I use everything. I don't want to have some compressor sitting there without doing anything. I'd rather have what I use and then borrow or rent or buy whatever I need.

"My monitors are Genelec 8050s with a Genelec sub, and Dynaudio BM15s with a Bryston amplifier. Having a sub is essential nowadays. The Dynaudios are very transparent. I had them in my previous studio, so I also kept them for continuity. I did a lot of work on the acoustics and the insulation in this room, and tuned the Genelecs to the room."

Following the death of Andy Fletcher, Depeche Mode are now a duo. (Photo: Anton Corbijn)

Fine Tuning

Salogni's mix process for Depeche Mode's *Memento Mori* album was rather different from other mix projects. "Of course, I was already really familiar with the sessions, because I had created them from scratch, and James and I worked on Pro Tools during the recording and producing. So there was minimal mix prep work involved for me.

"Normally I'd have tidied up the session, and I would have put things in order and coloured them, so I know my drums are at the top in blue, my basses are orange, and I don't have to distract my brain to go and find tracks. It means I don't need to lean into the screen too much while mixing. Screens are distracting devils!

"Also, James and I had regularly been doing rough mixes in California. We added many effects during tracking, and some during the rough mixes, and these effects were agreed upon by all four of us. The effects were final and had been printed. It meant that my main work here was to finesse the rough mixes, and make sure everything can co-exist, and be clear. Clarity is key for me."

As well as making the album sound coherent as a whole, Salogni also had to make sure that it would stand up against other contemporary releases. "First of all, this project was about itself, so I did not reference specific things. Having said that, I don't like things that sound retro. As someone who

works professionally in sound, I think it is a disservice to the technology that we have.

"Depeche had massive hits in the '80s, and everybody knows that these records sounded modern at the time. But we now have technology that allows us to reproduce frequencies that we were not easy to deal with back in the '80s. I believe it's important to make the most of the technological advancement of our time, unless a specific palette of sounds is the result of a creative choice.

"In addition, I want a record to sound good whether somebody plays it through AirPods, a small radio, or in a club, or in a stadium, or through a massive festival system. Quite a lot of artists I work with tour in clubs, so I need to make sure their songs will sound good in a club scenario. In my mix of 'Ghost Again' it means that the kick is upfront, and everything is clear. There is no mess or soup between the low mids and the low end. That is very important."

One Thing After Another

With the sessions laid out the way she prefers, Salogni set about the actual mix process. "The first thing I do is to mute everything. After that I unmute one thing at a time, solo it, and EQ it if needed. I then solo another element, and do the same, and then I will make sure both sound good together. I then do the same with a third element, and after that it is a process of adding tracks in one by one, and making them sound good on their own, and good in context.

"I'll be doing a lot of subtractive EQ at this point, to make sure things can coexist without getting in each other's way. It's a matter of shaving frequencies off one instrument if it's already covered by another, and then making sure you don't lose something. In a way, subtractive EQ is an anti-climactic thing to do, but when I don't do it, and turn things up loud at the end, and put on a compressor, the mix can fall apart. Whereas when you have done subtractive EQ, you can make the track very loud and present.

"I unmute things in the order of priority in the track. In one mix the first thing I unmute may be the lead vocal. I may unmute the vocal, and then the kick, and then the guitar, and then the bass, and so on. With 'Ghost Again', I started with the drums, because the track is based on a drum loop from Martin's demo, and then we built on that. I would have started by making the loop sound good, because that is one of the first things that gets you hooked in, and then the bass, which is quite driving.

"The bass comes from the Moog modular, and the drums are made up of different elements, including the loop, a Roland TR-606 drum machine, and the Eurorack Noise Engineering Basimilus Iteritas module. We also layered the guitar a few times. It was played by Martin, and we recorded it through two amps and a DI. The guitar also went through tape machines, the Korg MS-20 used as an effect, the Mu-Tron, a Dimension-D and an AMS RMX16, all outboard.

"Once again, when I mixed, these effects were already in the session. Because we already had so many analogue effects in the sessions, I did not send my mixes out to my Studer desk, other than just a couple of things. If a track was missing a bit of presence,

»

» and I knew I could get the best result from the desk EQ, I would send it to the Studer, and then print it back in. I also added a few tape recorder delays here and there, like a slap echo to the vocal in in 'Ghost Again'. But after these two and a half months in California, we had plenty of tape effects."

Drums & Bass

The process of "refining" the rough mixes involved a lot of automation on volume, panning, and a considerable number of plug-ins. "There were four plug-ins on the drums bus track: a Soundtoys Decapitator, a FabFilter Pro-Q 3, a UAD LA-2A and the EQ7 from Pro Tools. There's also a drums parallel, which is sent from the drums bus track, and it has the Soundtoys Devil-Loc and a UAD AMS RMX16. I was still adding distortion during these final mixes, I really rate the Devil-Loc plug-in for that. I also use quite a few UAD plug-ins, and have the UAD-2 Satellite Thunderbolt Octo Core.

"With the bass, I really like having a lot of it, and for it to be as clear as possible, so it sounds good on AirPods, mono speakers, stereo speakers, with or without a sub, speakers that are from today, and speakers that are from the '70s. I want a listening experience that is democratic. But I also want a listening experience that is the best that it can be. It means that I A/B my mixes on many different systems.

"In this session I have some Devil-Loc on the bass, as well as some Soundtoys MicroShift, because I wanted to stereo the bass a little bit. The mix setting on the MicroShift is quite low. The Devil-Loc creates some saturation and distortion, and I also added some presence with an EQ. Both aim to make sure the bass can be heard on speakers that don't have much low end."

Guitars & Synths

"After the bass and drums I would have brought in guitars and synths. We recorded each guitar part, with parts doubled, over several tracks: room mic, DI, amp 1, amp 2, Mu-Tron, AMS, Lexicon. For each part these went to a group aux track: guitars 1, 2 and 3. I had plug-ins on these aux tracks, like the Pro-Q 3, LA-2A and Valhalla Room [*reverb*]. These then went into a master aux track, which allowed me to unify the guitar sound.

"There are quite a few synths. There's an Elka Synthex, an ARP Solina String Ensemble, and there's a synth solo from a mixture of synths, including an Arturia PolyBrute and a Buchla. The panning effect is a stereo setting on the Elka — the effect happens as you play. I added quite a few plug-ins on the synths, including the Valhalla Vintage Verb on the Elka, and the UAD EMT 250 on an aux effects track. I like the Valhalla on the insert, because I can use it as a widener as well. It's a really good plug-in, pretty magic. I have the Waves Doubler on the Solina, which pushes it wide and brings out some shimmer.

"We ran the synth solo through many things in California, including the tape recorders and the Overstayer. During mixing I added a [*Oeksound*] Soothe 2, because all these synths, especially from the Eurorack, have a huge frequency range, and sometimes the high frequencies are too prominent. Soothe does a great job

Studio Zona is home to a quirky selection of synths and effects devices.

Soundtoys' Devil-Loc was used to add saturation both to the drums and the bass synth.

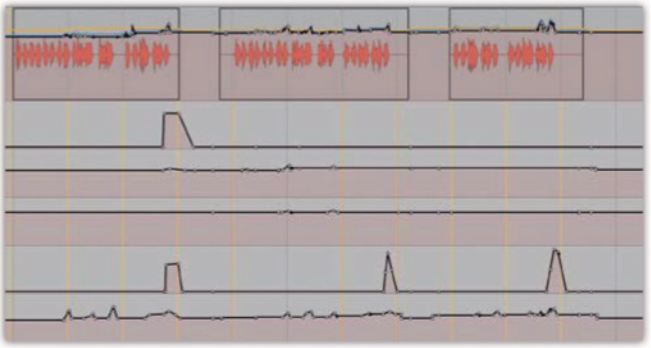
of attenuating high frequencies like that, without killing the presence of the sound. I also use the Pro-Q 3 on the synth chords to take out low end, doing subtractive EQ, so it does not get in the way of the bass."

Vocals

"There is a lot of volume automation on Dave's vocals, because I wanted to make sure the lyrics are clear. His vocal performance is great, and I wanted it to shine above everything that is happening, which is a lot. Sometimes the endings of his phrases have a little bit of volume automation, to make sure they cut through. I wanted his performance to feel very dynamic, and for the effects to support his performance.

"I automated the effects for different parts and sections, so that they all feel slightly different. A lot of work went into that. Plug-ins I used on his vocals include the Devil-Loc, Oeksound Soothe, and Valhalla Delay. The Devil-Loc can be quite subtle, and works well on vocals. I also have EQ, compression, EQ, de-esser, then another EQ, Devil-Loc, [*Soundtoys*] Radiator, Soothe, and EQ at the end.

"I wanted Martin's vocals to have their own space, which was different than Dave's, so they could each be in their own worlds. There are two tracks with Martin's vocals, panned slightly left and right, EQ'ed and then the Waves H-Delay on a ping-pong setting helps to spread them. The Valhalla Vintage Verb smooths out the repetitions. Then there is another EQ, which just takes down some of the low mids, so that the vocals are nice and clear. Finally, there is a send, which goes to a slightly different HDelay, and a slightly different Valhalla."



Extensive level and pan automation was used to fit David Gahan's vocal parts into the track.

Master Buss

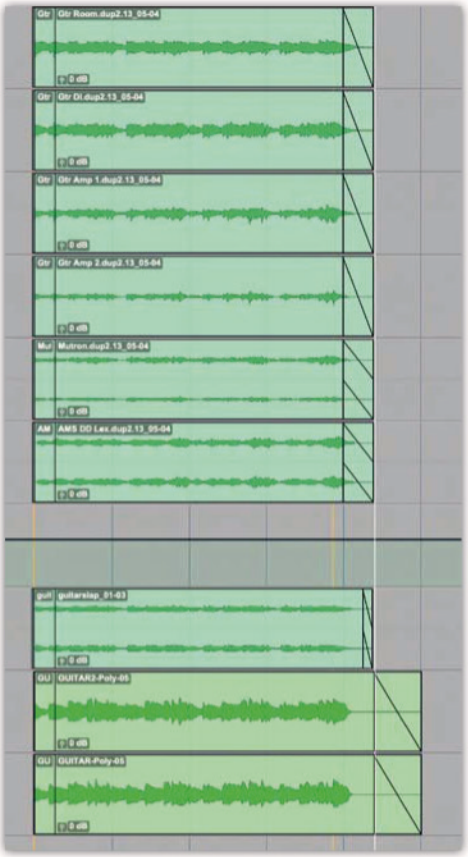
"There is the important process of making the mix sound good as a whole, which means that as a mixer I do a lot of work on the master. I will have EQ and compression, sometimes I will add saturation, sometimes I will play with the width and I will automate that, or I may automate the output of the compressor if I wanted one section to pop out. Sometimes I will add effects on the master.

"I mix with nothing on the master bus until I get to just over halfway into the mix, and then I complete the rest of my mix with plug-ins on the master. These plug-ins make the mixes feel like a record. The four plug-ins I have on the master bus in this track are my go-tos: the PSP Master Q2, UAD SSL 4000 G, UAD Chandler Curve Bender, and the Linear Phase Multiband.

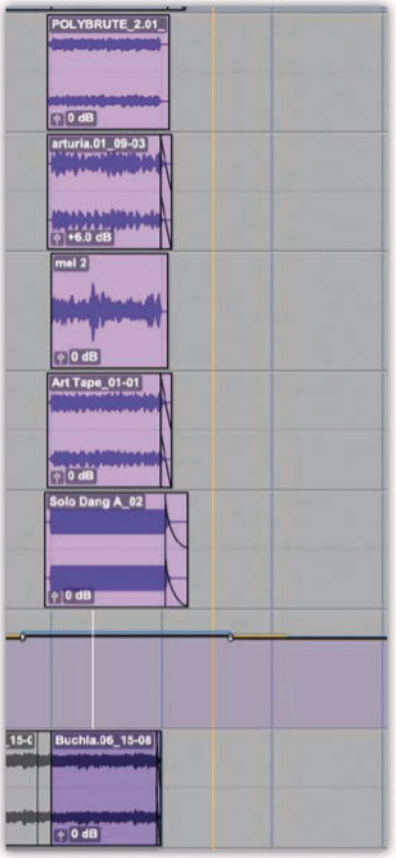
"The PSP allows me to EQ the whole mix, and to lower the inputs in case I need more headroom. Then I have the SSL 4000 G, for compression. I like using a little bit of compression as it helps everything to gel. I always use a slow attack and a fast release so I don't compromise any transients, which is very important for kick and bass. Then I have the Curve Bender, which is a very good and simple EQ. The real thing sounds good, and so does the plug-in. UA have done a great job with modelling it.

"And finally, the Multiband allows me to tackle frequencies that may be a little bit too prominent in certain sections, but not in others. I can compensate for any loss of volume by turning up the gain. I sometimes add in an EQ at the end, often with the Pro-Q 3, just for last-minute tweaks, to see whether I have still more space for bass or for top end. Of course the mastering engineer will look at that, but I want to make my mix the best it can be before it gets to them.

"Towards the end of my mix, I will also put a limiter on my master bus, because I want to know what it sounds like at mastered volume. I use the DMG Audio Limitless for this. I make the mix loud so I can address it as a track that is already mastered, and it may lead me to make any final tweaks. After that I take off the Limitless to print the mix, and I send that to the mastering engineer."



The guitar was multitracked through a variety of effects, all of which were printed to the Pro Tools session.



The synth solo in 'Ghost Again' was created from several layered sounds.

MICHAEL ROMANOWSKI

Mastering Spatial Audio

Four-time Grammy winner Michael Romanowski is one of very few people offering mastering services for immersive music.

MORITZ HILLMAYER

The term 'mastering' is rarely used in the context of immersive audio. Nevertheless, it's a service that Michael Romanowski has been offering for some time, and he was among the first to build an immersive mastering room, at his Coast Mastering studio. "That was four or five years ago, and the studio was recently nominated for a TEC Award for studio design," Romanowski explains. "We've been working with Dolby, Sony and other companies on their software, helped them develop the platforms, and were involved early on in the transition of Dolby from doing surround or immersive sound for film to music. I built my first surround room in 2001 and started mixing and mastering with height speakers in 2017."

Master Of Mastering

Romanowski's career followed the "old-fashioned way", as he calls it. Starting out in live sound, Romanowski moved on to recording and mixing shows

Michael Romanowski at his Coast Mastering studio. (Photo: Cheryl Alterman)

for records, then got into producing, mixing and eventually mastering bands in the studio. Although mastering has been his focus for almost 30 years, he still does mixing and even the occasional live show. “It keeps me fresh and informed about what’s happening, how people are mixing, what kind of tracks are coming in. And I still have fun doing it.

“I believe very firmly in the approach of having a different mastering engineer from the mix engineer. It’s helpful to have a second set of ears. It’s helpful to have somebody in a tuned environment looking at a 10,000-foot view rather than a microscope. Someone who sees how the music presents itself, not what it is. My job is to know how the music is, not what the music is. In stereo, that’s huge. You’re putting your heart out into the world, you get one shot at it. Once it’s out there, it’s out there. If you don’t put your best foot forward, people are going to know it, and why should they be investing their time in you when there are a million songs a day released across the world? You better present yourself as best as you can because that’s your only shot. So one of the core objectives of mastering is to be as translatable as possible across all the different platforms, physical media, streaming, LPs, etc.”

Immersive audio likewise offers many different listening options. “There are so many different ways people are listening to immersive audio right now, from sound bars to headphones, from home systems and TVs to computer monitors. And with multiple formats, mastering to me is absolutely essential, because there are so many places for things to potentially go wrong. Most people have tuned their listening environments to their personal preference. But that does not necessarily translate to the commercial market. So if you don’t have a second set of ears listening to that to know how the music translates, you could end up with compromised sound that may only be good in that one particular environment.”

Mastering is about more than just translation, however. “The other part of mastering is cohesion. A lot of records are mixed by several different engineers. If you don’t take care of looking at the body of work, the entire album, then the collection of songs just sounds like a playlist and not an album. That doesn’t do the artist justice. It’s about going through the entire album and making sure levels work between songs, tonality is the same between songs.”

When it comes to immersive audio, Romanowski prefers to not interfere with the mix itself. “When I get the files for mastering, I have access to the individual objects and channels. I could adjust each of them individually if I wanted to. But if I’m hearing something out of a particular object that needs adjustment, I’ll talk to the mix engineer first. If they can make an adjustment on the mix, then I don’t have to do something on the master that could potentially compromise other balances. If you’ve got this setup around you, and there’s an object on one side, it’s part of the balance of the tone. Now if you realise that it’s too loud or too bright, and you do something to fix it, that can shift the whole feel.

“If I have the chance to speak with the mix engineer, I consider it an opportunity. There’s a dialogue, they get feedback on the sound of the mix and the presentation and it just improves the final material so much. Then I can polish, which is what mastering should be: that last little bit to make sure that it’s translatable, and eventually creating the correct master for delivery.”

The Wild West

The early days of stereo were a period of experimentation, when engineers toyed with radical panning ideas. To Romanowski, the current situation in immersive audio production is comparable. “You’ll notice that with the first few stereo records, there were two camps. There was one that said: everything needs to come out of the centre, it just needs to be a little wider to feel like it’s bigger in front of you. And then there were those who said: let’s put all these instruments hard left and put all those instruments hard right, and sort of create this separate field. And some of both worked and some of both didn’t work. What the Beatles did was amazing, but those were throwaway mixes. They worked so hard on the individual mono mixes, and then for the stereo ones the guys wanted to pan this and pan that and have some fun with it. Listen to ‘Are You Experienced’ by Hendrix. It has this swirly guitar that sounds really cool on speakers, but when you put headphones on, it doesn’t work as well. So, some things worked, others didn’t, and we’re in that world with immersive audio right now. The rules are: Try it. If it sounds good, do it, if it doesn’t, don’t.

“So far, we as an industry have been mainly focusing on back catalogue albums, things that are already done. Everything so far has been for the most part recorded and mixed to come out of two speakers. Parts used to be written, arranged, played and recorded to fit in a very compact left speaker/right speaker wall in front of you. Now that we have all these walls to go around, the instruments can live in different places, which means they can function differently in a composition. They can add to, they can talk about, they can have dialogue, »

question and answer that can be totally separate, or they can be completely the same."

New Possibilities

Michael Romanowski views catalogue remixes as a way to ease people into the new format by refreshing music they already know, but he feels that the real creative potential lies in new productions created with immersive listening in mind. "The potential change in music production is going to be great! Hopefully, engineers are going to be questioning the use of close miking for everything to be in your face and over-compressed. It should be a creative choice. I'll be honest, I'm not a fan of most over-compressed music. It often just makes me want to turn it down or walk away. But now, with immersive audio, we have the opportunity to work with transients and dynamics, space and distance, near and far and all of these things. No longer does the mic necessarily need to be jammed up in the speaker of the guitar amp. We can have a sense of space and capture things with full waveform development. It's amazing how natural it sounds when we do these kinds of things. And I think immersive audio is going to change people's production techniques. I honestly feel so strongly about that, that I think it'll also change the way we make stereo records.

"I think there's a few things going to come from this. First, certain sounds were manufactured for specific genres. They are not real. Like kick drums in metal and hip-hop. They were created to fit into a temporal and dynamic sonic space. For example, when Metallica became prominent with those tiki-tiki kick drums and toms, it was a creative decision to have them in a higher frequency range, which allowed different energy to be in the lower portion of the spectrum. To me, there was an 'A-ha' moment when Tom Petty's *Wildflowers* came out. There wasn't a lot of low-mid energy. There were lows where the kick drum and bass were, and there was some upper midrange, but not a lot of low mids. That allowed for energy redistribution, it helped become a louder record than people were used to listening to, which all of a sudden became an exciting new sound in a particular way.

"But with immersive audio, we have the opportunity to go back to original sounds. If Metallica wanted a regular kick drum sound, now they can put that energy in a position in a 360-degree space, so they don't have to force it into a spot. They still may choose to do the tiki part because that might be the sound that they like. If people like what they have, even if it originated from restrictions that they had, would they still do the same thing simply because they like it? That's the big question: because you can, will you or should you?"

Making Space

It's not only artists who need to get used to the possibilities of the new formats. "The biggest learning moment for listeners is the change of sound on their headphones. Most people will be listening on headphones, surely, and they're used to hearing a centre image right in the middle of their heads. With immersive, the image on headphones is out there. That's a little jarring, because it sounds not as immediate, it's not as pounding. But if you take a moment and listen, it actually is more engaging, because it brings you into the music. When we look at production techniques, in order to get as much music to come out of two channels as possible, people have been compressing and compressing and compressing, getting rid of as much dynamic range as possible, so everything is just right in your face and flat and centre. When you do that in immersive, it brings the signal back to the centre of your head.

"The more that we do those things for loudness and compression, the more we take away what it is that we like about it in the first place. You need to work your production and engineering techniques to best suit this sense of space. Some people will say that they don't want to change the way they're used to working, they've always done it like that and don't want to change. Great, you don't have to. Keep mixing in stereo. Awesome, own it. Other folks are embracing the new possibilities, the clean slate, they love what they can do with immersive formats, so let them run with that. Let those who want to stay in stereo do that. Excellent. Intentional stereo mixes will always be important. There's plenty of people who want to explore, figure out and try something new.

"A possible new approach could be to not take each drum and compress it, and then run it to a stereo bus and then compress it and then do parallel stereo compression of that, and then run it into the master bus and then compress that. We don't have to keep doing that. And if you're willing to step back from that paradigm, I think that's where much of the potential success with this format may come from. Creativity."

Coast Mastering employs a mixture of Focal and Neumann loudspeakers. (Photo: Eric Rorer)

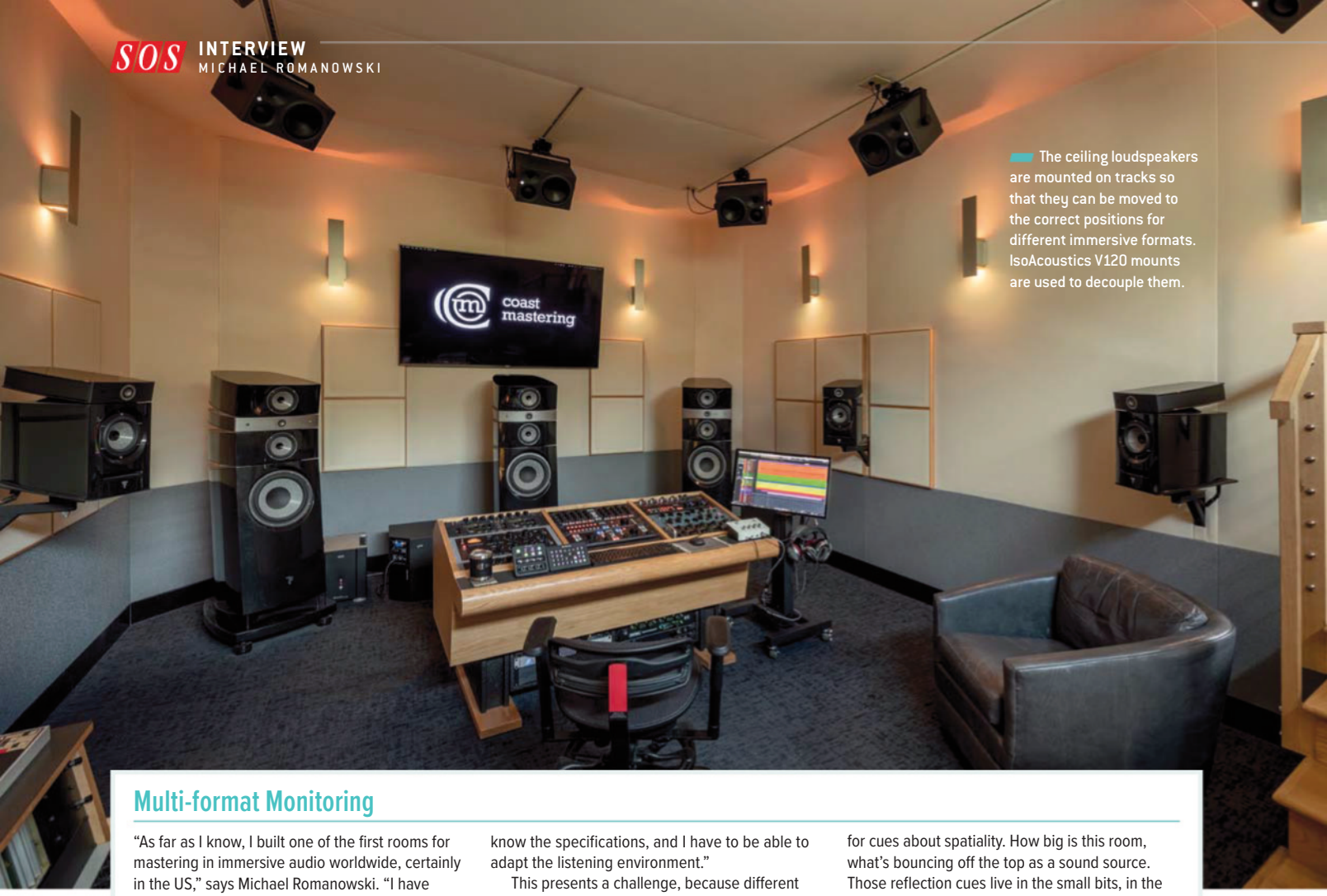
Romanowski senses that some engineers and producers struggle to wrap their heads around the possibilities of an object-based environment when they're so used to a channel-based one. There appears to be a certain reluctance to let go of old workflows, to leave behind the tried and tested ways of placing instruments in speakers in favour of placing them in a space. "I guess a part of

this is: how did they learn to become an engineer? What brought them here? Were they just part of a band and figuring it out to record their own material, and then figured they could produce other people, too? Or is it a passion to learn the best about sound and be fully engaged in the engineering processes? Is it somewhere in between? I find the folks that are curious and experimental are the ones that are embracing this."

From Stereo To Spatial Audio

Even among those who embrace the new formats, there are very different approaches: some use the three-dimensional space for effects, others for atmosphere. "The one side is trying to really respect the stereo mix, and just make it a little bit bigger, a little wider, a little taller, or just a little more, but still essentially the stereo mix. Other folks want to explore a little bit, get more creative with it. I've worked with both kinds of records. I mixed Bonnie Raitt's last record in Dolby Atmos and Bonnie was in the studio. I played her a whole bunch of stuff, classical, jazz, rock, just different things to give her a feel for it. And when we sat down and started getting to work on it, she realised that she's basically a five-piece band. She just wants to be right here, somewhat around you. A sense of space, but not like 'guitar part back here and keyboard over there', that doesn't fit.

"Respect the music as the artist intends, first and foremost. And if the artist wants it a certain way, then that's what you do. I also mixed the Fantastic Negrito record. Same kind of thing: I came in, sat down with him and played him some stuff. What does he like? What does he not like? Then my question was, how far to get away from the stereo mix. Because he's got a whole lot of sounds and designs in it. He looked over and said: 'I already have a stereo mix, do whatever you want.' It's awesome for an artist to be able to say 'I already have stereo. Why would we do the same »



The ceiling loudspeakers are mounted on tracks so that they can be moved to the correct positions for different immersive formats. IsoAcoustics V120 mounts are used to decouple them.

Multi-format Monitoring

"As far as I know, I built one of the first rooms for mastering in immersive audio worldwide, certainly in the US," says Michael Romanowski. "I have a hybrid 9.1.6 system here. Everything on the floor is Focal, from Stella to Diablo Evo. My ceiling speakers are Neumann 310s. But I also have the centre height speaker, and two on the floor for Sony. Those are also the 310s.

"I can work entirely in the box, or entirely analogue. And I can work with all of the current different formats, each having unique workflow and specifications: Dolby Atmos has beds and an LFE, Sony just has objects, same for Auro 3D. There is a handful of different formats out there and we know that there are more coming.

"If I'm mastering something in stereo, they might release high-res files, a CD, an LP, so they made three different formats. Not unlike this, somebody may have mixed in Dolby Atmos, but they also need to release the Sony 360 or a binaural format. So I have to be able to handle all formats as a mastering engineer. I have to know if it sounds as it's supposed to sound before it goes out. It's hugely important to me that I can adapt. I have to know the software, I have to know the specifications, and I have to be able to adapt the listening environment."

This presents a challenge, because different immersive formats specify different speaker layouts. Romanowski solved this problem by mounting some speakers on Unistrut tracks so that they can be moved according to the specifications of the required format. "If I'm doing Dolby Atmos, I have them set a particular way. If I move to Sony, I have a particular way. When I'm doing Sony, for example, the two speakers up on the side come down to the floor. I've got them positioned and routed so that I can just bring them down. So I'll adapt the room listening to the projects I'm working on."

Romanowski's solution required special attention to the often under-recognised problem of speaker isolation. "They're hanging from this track, and the speaker vibrations can get into the track and transmit it to the other speakers."

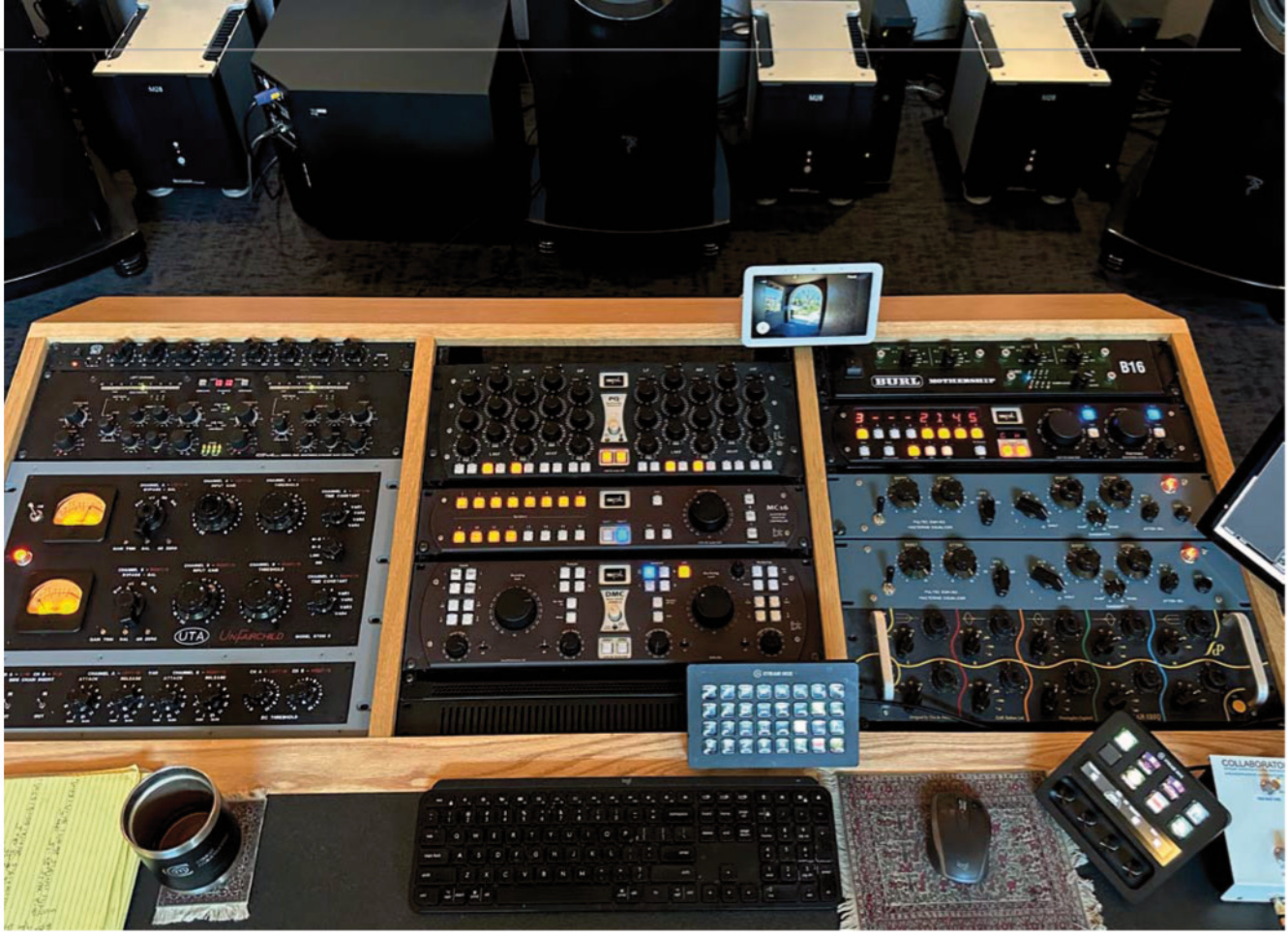
This is a particular issue with the ceiling speakers, the main role of which is to provide subtle spatial cues to the listener. "To me, the benefit from the ceiling is that in most immersive music, the height speakers are really used more for cues about spatiality. How big is this room, what's bouncing off the top as a sound source. Those reflection cues live in the small bits, in the fine resolution and in the detail. They can easily be obscured, by jitter problems, if it's due to digital audio, or bad amplifiers; a whole host of things can get in the way. Same thing with the speakers being hung. The imaging can be compromised because they're not isolated, one may be getting a little time-delayed signal off the other, so now the picture of the ceiling is different."

Consequently, Romanowski is an enthusiastic advocate of IsoAcoustics' speaker mounting products. "The speakers are decoupled with IsoAcoustics GAIAs for the speakers on the floor and the new V120s all the way around, holding up the ceiling speakers. The idea of IsoAcoustics in general is to decouple the energy coming out of the speaker from whatever surrounding device it's on. What the V120s are doing for me is isolating that speaker from that track, just like the other ones do from the floor. The speaker is vibrating by itself and doing what it's supposed to be doing, rather than transmitting that energy to another source."

» thing in a bigger space, we already have that, let's do something else.' I like that creativity to be able to go and have fun with it. Not every record can do that. Not every classic record is meant to be brought back. Not every record needs to have movements and shiny objects and stuff floating around. I think one of the biggest things that we can do as engineers and producers, for artists, as service providers, is to help the listener listen to the music, without paying attention to the process that the record was created from."

The Listening Process

Naturally, Michael Romanowski puts a lot of emphasis on the quality and quantity of listening when working on a record. "Every decision I make is because of what I'm hearing. If I can't hear it correctly, then every decision I make is wrong — or potentially wrong. Every step from there is developing two opinions based on what I'm hearing: I'm hearing this, and I think it could be this. I'm hearing that this is too bright, I'm



Michael Romanowski's mastering console hosts a selection of top-flight analogue processors from UTA, GML, Burl Audio, SPL, Pultec and EAR. (Photo: Michael Romanowski / Coast Mastering)

feeling that it could be less bright, and then the action is to adjust whatever it is that makes that too bright — whether it's a guitar part, a vocal or a whole mix. The process is listen, observe, decide, and then act and make the change.

"I have a very particularly tuned room. The acoustic design is by Bob Hodas. So I know what the sound of the room is. Because I must know how it translates, this goes back to the task at hand: translatability. If I don't know what it sounds like here, I'm not going to know what it sounds like in cars and in headphones and all sorts of things. As the saying goes, if you're on the wrong train, every stop is the wrong stop. Every decision I make is based on my perception and if my perception is off, so are my decisions.

"I don't choose what gear or what process I'm going to use until I know what it sounds like. Some engineers start with a mix bus preset that has a couple of compressors and EQs and stuff like that on it, I think that's totally wrong. I think that's the worst thing you can do. To me the analogy is: you don't start with a cup of salt and decide you're gonna make soup, and you have to add enough ingredients to make the salt work. That doesn't make sense to me. Make the mix work, and then season it to taste at the end.

"Presets are places to start. They're the first step, not a destination. Let's just say you have a preset 'rock number six' for a compressor for a kick drum. You don't know the beater, the head, the drum, the room, the microphone, the player, or the song they're playing. There are so many things you don't know that a preset can only be a place to start. I'd hazard a guess that anybody who works with prepared chains will go listen to a chain, then another chain, and another, and then choose one. But now within that chain, there are EQs and compressors and things to adjust to make that preset work for the material at hand."

Romanowski himself works mainly with analogue processors, and chooses which ones to use only once he's decided what work needs to be done. "I do everything in the analogue domain, for the most part. A handful of different compressors and EQs, of which I'll choose the ones that work best for what I need to do. I do as little as I can, frankly, because I don't want to destroy the music. I'm not here to put my flavour on. It is not about my sound, it's about the artist's sound. I'm here to make them sound more like them when they leave, than they did when they came here. My name is small and buried deep in the fine print, if at all; their name is front and centre on the front cover. It's about the band or the artist and their songs.

"We have 12 or more channels that we have to deal with in immersive audio. There are so many places where things could go wrong and not have the music be represented as the artists intend. So, I think mastering is even more important in immersive audio than it is in stereo."

> Michael Romanowski: "With immersive audio, we have the opportunity to go back to original sounds."

How I Got THAT SOUND

Tom Lord-Alge
▪ Live 'Lightning Crashes'

JOE MATERA

Three-time Grammy Award-winner Tom Lord-Alge has an extensive list of multi-platinum landmark albums and singles to his credit, including Blink 182's *Enema Of The State*, Steve Winwood's *Back In The Highlife*, the Rolling Stones' *Bridges To Babylon* and Weezer's *Maladroit* to name but a few. Asked to pick a favourite sound, Lord-Alge highlights the guitar on Live's 1994 hit 'Lightning Crashes'.

"I came in as the mixer as the album was recorded by Lou Giordano. On the recordings, with the guitars there were anywhere up to three to four different microphones on four different amplifiers all going at once, so I got to choose between what I would use."

Feel The Noise

"On 'Lightning Crashes', the first thing I noticed was that the guitar amps were really noisy. The song starts out with the guitar volume turned down, so it's kind of a clean sound and then it builds in intensity to the roar in the chorus. For the beginning part I just chose whichever mic and amp had the least amount of noise, because if you listen to the beginning of that song, the first guitar has a lot of noise going on. This was all done on analogue tape, as at the time it was right about when Pro Tools and DAWs were starting just come in.

"When I finally got the balance that everybody liked, Jerry Harrison, who produced the album, listened to it and said, 'Give that to me and let me record it into Pro Tools and let me see if I can denoise it,' as he had an early version of the software. So, it was my first introduction to Pro Tools too. And it seemed like it probably took him forever, trying to denoise the intro to that song. Then the band and I listened to it again and we were like, 'You know what, we like the noise!' So, we ended up going with the version as recorded, with all the amplifier noise.

"When the song starts it's just one microphone, and as the song progresses you can hear me start to feather in the other amplifiers, especially around the second verse, where you can hear another amplifier come in with some chorus effect on it. I just kind of mixed the track as I went along and feathered everything in."

The Last Resort

"I mixed the track at Music Head Studios in Lake Geneva, Wisconsin, which had been built in what used to be a Playboy resort. It was a big hotel and had like a convention centre attached to it. By the time we got there it was ready for the wrecking ball, but the studio was still operational. The studio had a great recording room and, on that song — and album in general — I miked up the recording room and set some speakers up in the recording room. Almost all of the ambience on that track was me just sending audio out to the room.

"It worked out well for the guitars, but also the vocals and particularly the drums. I did add some flanging but back then, I was doing the old-school tape flanging. I printed the instruments onto half-inch tape as a stereo mix, then would fly that back into the multitrack and just slightly vary the speed up and down to get the flanging effect so you got that full effect. That's the thing with flanging, you can't get it to go before the sound happens, otherwise you'll never really get that kind of jet engine sweeping sound that you would do if you did it the old-school way.

"I never would have chosen it as a single, so hats off to whoever decided to release it as a single, because it really was the song that defined that album and really made that album explode and sell well."

Hear The Sound

- https://open.spotify.com/track/5vVmnv4EN3X2VBa5Fo3UFM
- www.youtube.com/watch?v=xsJ4O-nSveg

SOUND ON SOUND

PODCASTS

Compression And Expansion

Seven Practical Examples

Eddie Bazil demystifies Compression and Expansion with a detailed explanation of their uses which he demonstrates using plug-ins through seven practical examples.

Daniel Moyler

Breakthrough Producer Of The Year Nominee

Daniel Moyler is a freelance mix engineer who was awarded Engineer Of The Year at the MPG 2020 Awards and won a 2021 Grammy for his engineering work on Dua Lipa's album Future Nostalgia.

The Anchoress

Self Producing Artist Nominee

Catherine Anne Davies, also known as The Anchoress, is a songwriter, multi-instrumentalist and producer. She is nominated in the category of Self Producing Artist at the Music Producers Guild Awards 2023.

Francine Perry

Rising Star Nominee

Francine Perry is an artist, producer and mix engineer. In this interview she talks with SOS's Sam Inglis about her career so far and being nominated in the Rising Star category for her work with Beth Orton, Erasure and HAAi while based at Mute Studio.

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SAMPLE LIBRARIES

Spitfire Audio

Abbey Road Orchestra: High Percussion

Plug-in Instrument

★★★★

The second instalment in Spitfire's new modular series is a direct continuation of the first. Abbey Road Orchestra: High Percussion picks up the thread of its predecessor, with distinguished percussionist and sole performer Joby Burgess patiently working through a large collection of percussive items in the hallowed acoustic of Abbey Road's Studio One. This time the drums are smaller and the overall sound palette generally lighter (though still powerful) — if you desperately seek the earth-shaking rumble of cinematic bass drums and taikos, you'd do well to consider ARO Low Percussion, reviewed in *SOS* January 2023.

The new High Percussion library is divided into 17 main presets, for which I'm happy to provide a quick overview. This being a symphonic project, it's no surprise to find a trio of orchestral snare drums: the first, a vintage 15-inch model, is blessed with a fabulous crisp, beefy tone, which to my ears is perfectly pitched. However, if you want a tighter, high-pitched delivery the library's two piccolo snares both pack a punch, with a large field drum also on hand for low, doomy 'executioner's drum' hits.

Moving into contemporary areas, a set of four octobans (a single-headed drum with a short sustain) and five clattering rototoms introduce ear-catching semi-pitched sonorities, while Latin staples bongos, mini-bongos and congas sound beautifully clean and transparent in the studio's reverberant acoustic. High and low timbales are also included, but lack the ferocious high-pitched clang one expects from the instrument. World drums are represented by a single darbuka Arabian hand drum, an African djembe and a miniature Japanese shime-daiko, the latter's colourful hits a personal favourite.

Another highlight is the log drum, a hollow hardwood box with six pitched 'tongues' — played with rubber mallets, it's an ideal timbre for light, bubbling percussion patterns. I also loved the pristine ambience of the temple blocks and wood blocks, but struggled to achieve an emotional connection with the plastic bucket hits. The collection is completed by nine so-called 'toys' which provide some essential percussion textures: six pairs of claves, pop-style shakers, castanets, guiros, caxixi basket shakers, maracas, ratchet, whips and the mighty vibraslap, without which no percussion collection is complete.

As before, instruments are deeply sampled with up to 11 dynamic layers and 16 round robins, including discrete right- and left-hand performances mapped for double-handed keyboard playing. Played with a variety of beaters, hands and brushes and recorded from 16 positions via a forest of microphones, the articulations comprise single hits, rolls and 'snare off' variants. This 98GB collection runs exclusively on the Abbey Road Orchestra plug-in and maintains the high quality of the first ARO release; however, considering the collection to date still lacks timpani, cymbals, metals and tuned perc, the price tag seems rather high. *Dave Stewart*

$449

www.spitfireaudio.com

Sampleson

Sandstorm

Plug-in Instrument

★★★★

I enjoyed Sampleson's Alienize when reviewing it in the January 2023 issue of *SOS*, so I was more than happy when their latest offering — Sandstorm — arrived in my in tray for review.

Like Alienize, Sandstorm is a virtual instrument running as a VST/AU plug-in (or standalone) via Sampleson's own engine/UI and includes the intuitive mouse gesture control system for sound modulation. However, the sonic target here is somewhat different as Sandstorm provides an easy-to-use platform for creating evolving soundscapes and sonic textures... although, as we will see in a minute, there are some melodic possibilities available also.

The concept is straightforward; the UI offers you 17 sound sources — a combination of three granular synth engines, two wavetable engines, eight sample-based layers and four other synth-based layers. The sonic flavour of each source is indicated by its abbreviated label — Str, Bas, Pad, Tub, Wind, etc — and you create a sound by simply blending your chosen combination of these sources. However, you can also modulate the blend by linking the level of each source to either the left/right axis of the upper part of the UI (where you then drag with the mouse) or to a MIDI CC number (such as your mod wheel). A good set of presets gets you started and demonstrates the possibilities.

While each sound source is essentially fixed in overall character, clicking on the small pencil icon for any source opens a compact set of additional controls that can shape the sound. For some layers this provides attack, release, reverb and pan controls. However, for others, it can be different options such as timbre (a sweepable filter), while the two sub layers offer LFO options and the three textures have a selection of different sound types you can pick between. And, while most of the layers produce soundscape-style elements, the arp layer provides a simple four-step arp sequence that can produce some interesting melodic content over the top of the more textural sounds. It's a shame that this layer doesn't also offer a wider range of sound types to pick from, although it does include a timbre control so you can change things up to some extent.

While the overall concept is, therefore, straightforward, by mixing and matching between the layers, you can coax a surprisingly diverse range of sounds from Sandstorm. What's particularly impressive, however, is how user-friendly the modulation system is to use. This makes it very easy to customise your soundscape/texture and give it a sense of motion. For composers looking to add tension to a scene and to 'hit' the key moments in the action, this works very well.

OK, so Sandstorm may not be the be-all-and-end-all of soundscape texture creation, but it sounds very

good, has enough sonic options to keep things interesting, and offers a very straightforward modulation system. For the budget-conscious composer, it's also accessibly priced. Well worth auditioning. *John Walden*

$59.99

www.sampleson.com

Inlet Audio
Clarinet Drama Textures
Kontakt Instrument

★★★★½

Emerging company Inlet Audio bill themselves as creators of "artist-driven sample library products". Here they add an intriguing new effort to their (thus far) string-dominated catalogue: Clarinet Drama Textures, built in collaboration with Chicago-based clarinetist and composer, Brandon C Stanley. The instrument (compatible with the full version of Kontakt 6 and beyond) allows users to pick any two clarinet textures from a library of 15 available textures, blending and shaping them with onboard tools to create pads, evolving soundscapes, and all manner of musical textures. All of the samples are created from a clarinet section playing various textures and extended techniques, making it a fun aleatoric instrument providing a new timbral palette for composers.

Diving straight in, I loaded a 'play and sing' texture into the A pane on the left side of the instrument, and Growl textures in the B side on the right. From there, each side has a Main and FX tab to individually modify the loaded sounds. The latter allows the user to mix in one of five onboard reverbs, from a small home patch to a moody plate, standard chamber, hall, and cathedral settings. It also provides a separate and character-filled delay module, moving from warm diffusion and vintage delays through a sparkling bright modern delay. It's in this delay setting where the fastest musical results in this instrument can be crafted, from subtle repetitions to immense eternal textures as the feedback is turned up.

While combining different sounds in the instrument is intuitive, sometimes taking the same sound and doubling it with different effects yields truly organic and seamless textures. For instance, I took the Growls patch — a moody, slightly edgy sample with lots of repeated key noise — and doubled it, using a hall reverb with an extended tape delay on the A side, and a smaller reverb with a slight diffusion delay on the B side. Rising from near silence to a forte dynamic with the mod wheel and then back down again, the result began as expected. Yet after a few seconds the sound doubled up on itself in the delay, and resulted in an extended esoteric piece of music that would not sound out of place in a Pierre Schaffer or early Steve Reich tape piece. Automation in the FX area became a must, though a dedicated panic button would have been a welcome addition. The front page of each side includes a simple envelope control, a standard filter (cutoff, resonance and bandwidth), and controls for panning, tuning and volume.

Clarinet Drama Textures certainly captures the unique warm characteristics of the clarinet, resulting in a highly original product capable of lending fresh timbral direction to your sound design. It is an intriguing creation, worth trying and at a very friendly price. *Mark Nowakowski*

$29

www.inletaudio.com

Void & Vista
Strands
Kontakt Instrument

★★★★★

If you have a passion for composing for film/TV contexts, or for more experimental song-based production, you may already be spoilt for choice when it comes to sound sources. However, as the cliché goes, the ideal number of virtual instruments is always N+1. If this applies to you then Strands — the first product from Void & Vista — may well be worth auditioning.

Described as a cinematic instrument and based upon a selection of sample-based bowed and electronic sound sources, Strands is presented within a very stylish Kontakt front-end. The core soundset is built from 32 sources (2.5GB in total) split into three categories; Bows, Effects and Waves. The Bows provide raw acoustic and electronic instruments and have clearly been sampled to capture the expressive capabilities of each source. The Effects group provides a set of heavily processed instrument and synth sounds, while the Waves group provides some purer synth and noise options. The UI lets you layer any two of the sounds as the basis of your preset and to 'morph' the blend of these.

There is perhaps nothing too radical in this basic concept, nor the 'organic-meets-processed' type of sounds, although the latter are undeniably good and full of sonic details. However, where Strands really steps up is in performance and modulation options then offered to draw out the full character of the underlying sounds. The UI's clever and stylish design makes this something of a joy to exploit. For example, clicking on any of the seven performance control labels opens up additional control sets related to that feature, providing you with access to ADSR envelopes (for amplifier, pitch and filter), filter configuration, 'motion' (two LFOs), options for additional voices (with detune and stereo width controls), lo-fi processing features, 'character' (essentially a three-slot multi-effects option) and a very effective delay/reverb. There are also slick MIDI options, making it easy to configure hands-on control of the key elements of the modulation features.

As a good number of the presets ably demonstrate, Strands is perhaps more of a sonic fit for darker, mysterious scoring duties (think horror, sci-fi or gritty modern crime rather than rom-com or period drama). However, if you should so choose, Stands can provide pretty much every element of your score including topline melodic sounds, percussive elements, rhythmic pulses and sound effects to go alongside those evolving textures/soundscapes. The UI is beautifully done and matched by the equally impressive underlying sounds. My only qualifier is that all that sonic goodness does require a fair amount of CPU grunt, but it's undoubtedly worth it; Strands sound fabulous and would grace any project, regardless of how 'Hollywood' the budget might be. It's also competitively priced and is well worth considering if you are looking for a suitable '+1' candidate to add to your scoring toolkit. *John Walden*

£129

www.voidandvista.com

Audio examples of this month's libraries are available at www.soundonsound.com.

How do I record taiko drums without distortion?

Two nights ago I had to record a Japanese taiko drum ensemble. I was using two DPA 4011 mics, but I got a fair bit of distortion even though my recording levels never reached 0dBFS; my actual peak was -3dBFS. Why was I getting this distortion/compression, and how can I mitigate it in the future? It looks like I'll be doing this again soon!
***SOS* Forum post**

Hugh Robjohns, Technical Editor
You say you were recording an ensemble with two mics, which implies a stereo array and therefore not close miking, which obviously reduces the SPL at the mics significantly, but even so the transients would be big. So transient overload in the mic impedance converter is a possibility, and for that kind of gig selecting the pad would be a wise precaution. A high-quality capacitor mic will capture brief transients which dynamic mics would round off, and which some meters won't show.

The next and probably most likely possibility is overloading the front end of the mic preamp. Sensitivity of the 4011 is 10mV/Pa, so if the transient peaks reached 120dB SPL at the mics, which is plausible, the output signal would be around -12dBu, which is approaching line level and might be getting a bit hot for some preamps, especially if there's a chunk of gain dialled in. Depending on the circuit design it's often possible to overload the front end of a mic preamp even with the gain at minimum. Input pads or inline attenuators are the saviour here, although the mic's internal pad may be sufficient to keep the mic signal within practical bounds.

Finally, sample-peak metering and intersample peaks within the DAW could be the culprit. Sample-peak meters commonly under-read on transient-rich material, often significantly, particularly if working at base sample rates. Under-reads of 6dB or more are not unusual. When replaying your recording it might be worth pulling the master fader down so the DAW output meters indicate peaks of -10dBFS just to see if the distortion is due to intersample peaks hitting the D-A converter in the replay chain.

The sensitivity of the excellent DPA 4011 mic means that loud transient peaks at 120dB SPL, which could be the case with taiko drums, would be output at nearly line level.

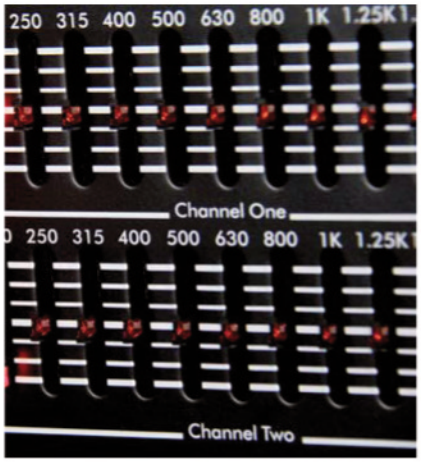
But if the distortion remains, it was a problem with the recording chain overloading somewhere — there is a declipper module in iZotope RX which works very well, and will probably salvage your recording to an acceptable degree. Personally, when recording anything (and relying on sample-peak meters) I get very nervous if peaks reach -6dBFS and I aim to keep them around -12dBFS or lower. Peaking to -3dBFS is just asking for trouble... With strong transient-rich material I'd be allowing even more headroom with peaks to around -15dBFS or so. The average signal level will be much, much lower than the transient peaks.

How do I figure out what's causing feedback?

We've just bought a Behringer Xenyx 1222 mixer. When setting the gain, we're not getting much past 40 percent on the mic channel before we start getting feedback. The mixer has feedback detection but none of the equalizers are indicating any feedback. We use the Hall effect on the mic channel but the feedback occurs without speaking into the mic, and this also occurred when the effects are off. I've seen a YouTube review that states using the compressor can cause feedback. Does it? Any suggestions on what the cause could be and how to eliminate it would be welcome!
***SOS* Forum post**

Hugh Robjohns
Feedback happens when the system loop gain exceeds unity. In other words, the mic picks up the sound from one or more PA speakers, either directly or after being reflected around the room. The signal from the mic is amplified by the preamp, through the mixer, and then by the PA amplifier (or active PA), before being emitted by the speaker and getting back to the mic. That's the loop.

If the sum of the gains in that loop add up to more than 0dB overall, the system will 'howl-round'. The preamp and power amp add the gain, while the speaker's inefficiencies and directionality, the mic's sensitivity and polar pattern, and acoustic losses through the air introduce attenuation.



While a graphic EQ can be used to reduce feedback at specific frequencies, it's better to break the loop by preventing/reducing the amount of sound reaching mics from the speakers.

Because the mic's frequency response isn't flat (and neither is that of the PA speakers), there will be slightly more gain at one specific frequency than others, and that's why howl-rounds occur at that pitch.

A graphic EQ is often used to reduce the gain at the pitch of the howl-round, which will kill it, but if you then increase the overall system gain it will howl-round at a new, different frequency where the overall gain is more than 0dB again! Skilful setting of a graphic can usually achieve 6dB or more extra gain in the system than without, but it's very easy to degrade the system sound quality with too much of this 'feedback correction'.

The best way to prevent/reduce feedback is to reduce the amount of sound reaching the mic from the PA speakers and/or floor monitors. You can achieve that by making sure the PA speakers are placed well forward of the mics, and/or using PA speakers with well controlled sound dispersion. Also, aim the null(s) in the microphone's polar pattern to reject as much sound as possible from the floor monitors/PA speakers — and avoid using omnidirectional mics!

Compressors don't cause feedback *per se*, but poor use of them can. Most have a 'make-up gain' facility, which is intended to maintain peak levels while raising quieter elements in the signal, but also adds to the overall gain in the system, and as I've explained, that's what causes howl-rounds. Compressors can be very useful tools for reducing peaks, but only if configured so as to make loud stuff quieter, not to make quiet stuff louder!

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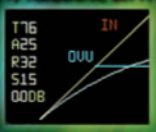

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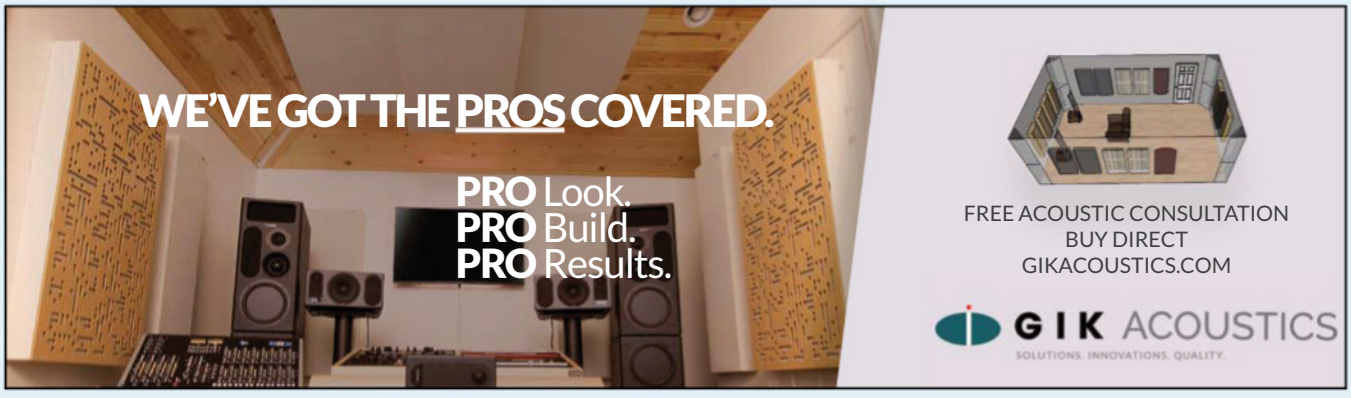

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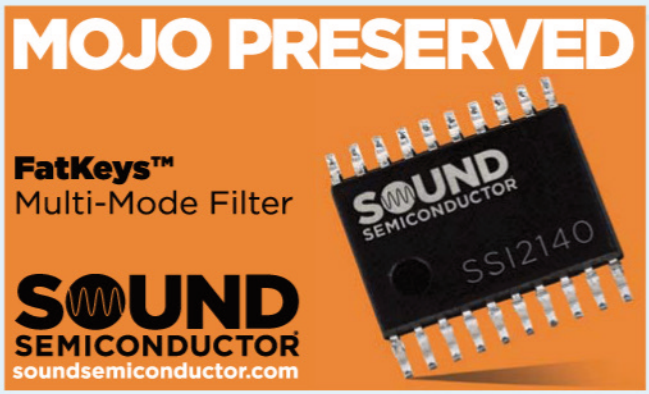
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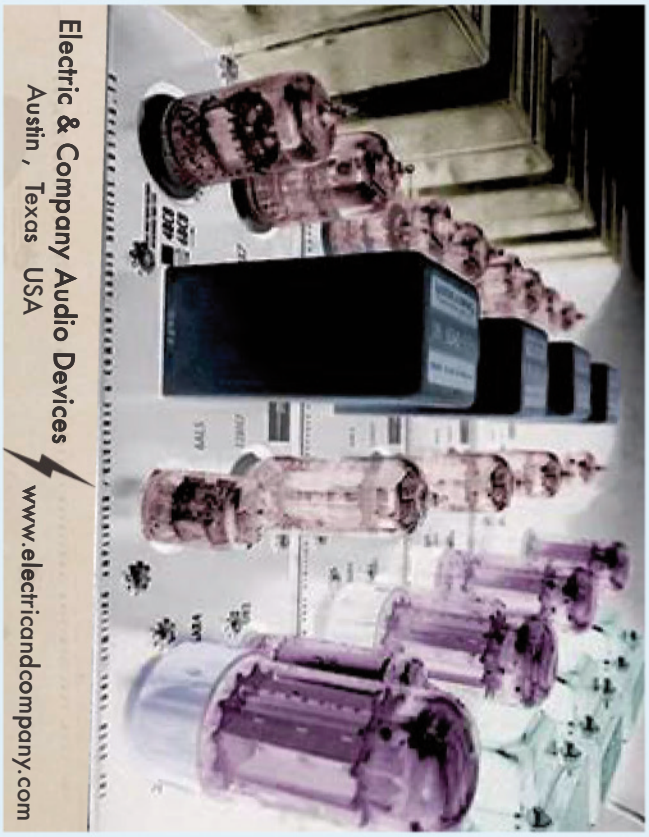
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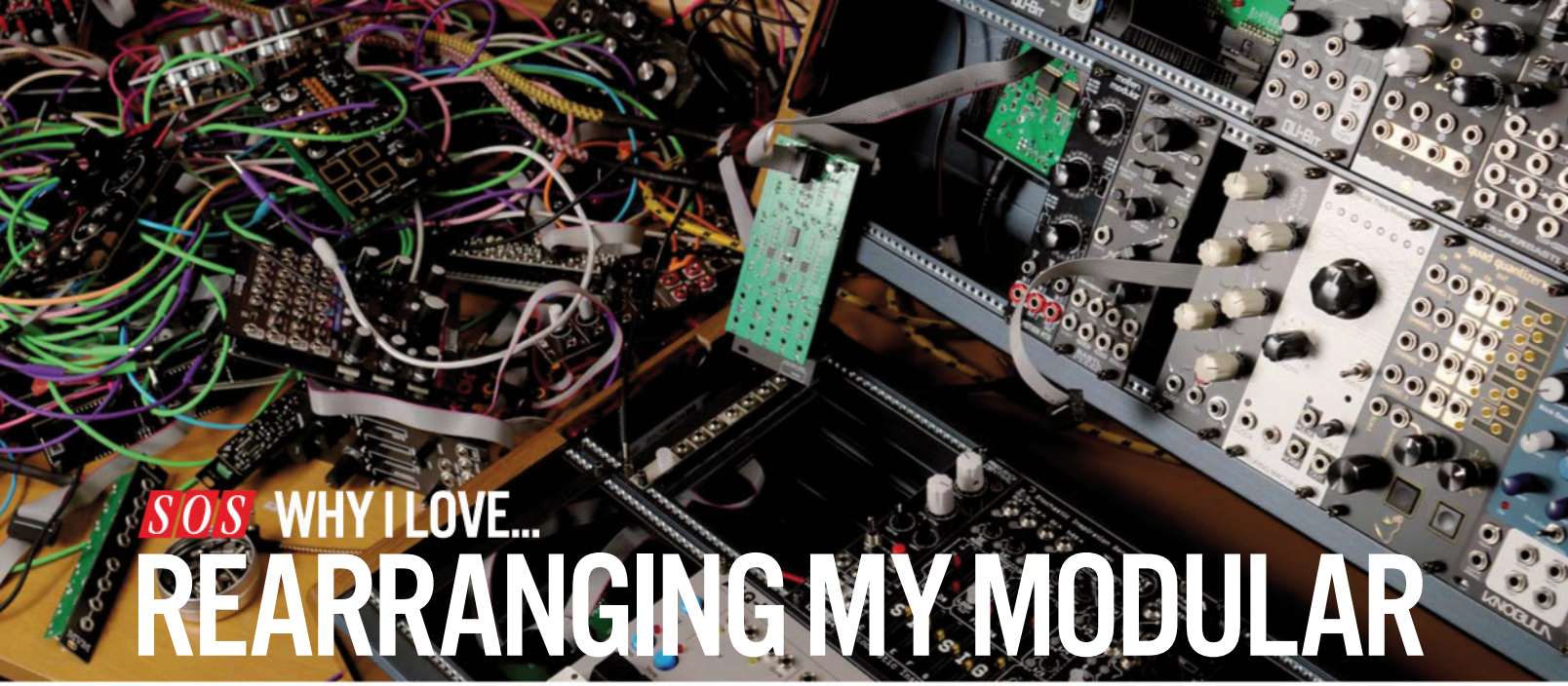

SOS WHY I LOVE...

REARRANGING MY MODULAR

ROBIN VINCENT

There's a running gag in modular circles about how someone has just completed their system. It's often accompanied by a beautiful photograph on Instagram of a modular synthesizer captured and frozen in the time of its consummation. What we know is that in a few days, maybe a week, other modules will start popping up out of drawers and things will change. For some this is a point of frustration and damned uncertainty. For me, I absolutely love it.

I remember the satisfaction of filling that first row and doing so with the exact number of HP. But I was already in the process of acquiring a larger case and so I knew it wasn't going to last. Even now, with an abundance of enclosures, nothing stays the same for long. I wouldn't call it a compulsion or an unhappiness with the way I set it up last time: it's much more intentional than that.

Every time I come to play with my modular, particularly when preparing for a gig, I go through the process of constructing the instrument. My mind scans through and examines the logs of my module collection seeking out moments of recognition and inspiration. I ask myself: What am I trying to do? What do I want to sound like? How can I replicate that thing I saw on Instagram the other day, or that thing I did last week? In that case I'll need this module, that module, a sprinkling of those utilities, and my currently active case gets enthusiastically unpatched and disassembled.

While my brain is busy making connections my hands are enjoying rummaging around in the case, passing through cables and handling the modules. There's something in the construction of the modular instrument, something in the putting together of these elements that is enormously satisfying. It's like the improvisation of the music-making is extending to the building of the instrument. I've also discovered that it's not only the changing of modules that sparks inspiration, the reshuffling of modules can do that too. On many occasions I've swapped a few modules around and found that it does something marvellous to the way the signals are flowing, and an unimpressive patch suddenly comes alive.

Then there's the puzzle element of wanting to fill a case without leaving any gaps, while keeping related modules clumped together. I can casually discard perfectly decent oscillators and filters to make way for an alternative that's 3HP larger or smaller just for the satisfaction of filling a gap or completing a row. Damn those odd number width modules!

The downside is that I have half-scavenged cases strewn about the place. Modules droop their power ribbons off the edge of shelves and mingle about on surfaces with pools of patch cables. But once everything is in its place, you have the joy of switching it on and beaming as the LEDs bounce, digital things boot and your speakers boom with anticipation. Hopefully there's still time to make some music.

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